USER DOCUMENTATION

Nor-840

Real Time Analyser

A dual channel real time analyser designed to meet the most stringent demands. Alongside with the more traditional 1/1 and 1/3 octave band analysis, the analyser can also do (optional) fractional narrow band octave analysis, FFT, sound intensity measurements, reverberation time calculations and measurements using maximum length sequence.

The analyser has a bright colour 10" LCD screen and the operator accesses the setups using dedicated front panel keys to open logical menus. All setups are selected using only single-level menus – so there is no need to navigate through multilevel menus.

CE

VOLUME I

Using the Nor-840

Norsonic

Using the Nor-840

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Thank you for choosing Norsonic! The real time analyser Nor-840 has ben designed to give you many years of safe, reliable operation.

Your approach to the Nor-840 documentation depends on what you want to do and how much you already know.

The manual has been divided into eleven sections plus index. Each section provides different information. Some places information has been copied from other sections (but adapted) to let you have all the relevant information there and then, thus avoiding unecessary "page-riding".

Depending on your requirements and your familiarity with technical acoustics as such, you may find that you use some parts of the manual often and others not at all.

The manual describes a fully equipped instrument. Your version may not have all the described optional extensions installed. The extensions may, however, be installed as a retrofit at any time in any analyser. The idea of offering future expansion is a fundamental part of our business concept.

The manual's structure is well-suited for reference purposes, but beginners should note that the order of appearance of the manual's topics is neither arbitrary nor alphabetic.

Instead we have sorted the topics in an order reflecting the natural flow of work when dealing with a specific task. For example, the section *Making Level Measurements* starts with a discussion on general measurement aspects then proceeds through the complete measurement preparation procedure (in the recommended order), before an outline is provided on how to actually make the measurement and on the display tools you have at your disposal.

For your convenience we have prepared an extensive index – please use it!

We consider the section *Basic Concepts* as the most important part of this manual, because it explains all the fundamentals, from transducer connections and battery handling to the principles of menu handling and parameter setup.

Consequently, if you are going to read just one section in this manual before you start using the analyser, we strongly recommend that this should be the *Getting Started* section.

The rest of the manual can then be consulted whenever required.

Observe the analyser's built-in on-line help function. For each parameter-field there is an associated help box. This context sensitive box is produced by pressing the HELP key and contains the valid ranges for the parameter together with a short description of its function.

Our main objective with this manual was to address your goals and needs. Please let us know how well we succeeded!

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Getting Started

Your portable Nor-840 real time analyser is easy to use, once you have understood the operating principles.

All setups and the control of the analyser are made by pressing dedicated keys which in turn open dedicated menus.

All these menus are single level menus, so you do not have to navigate through multi-level menus.

There is no special initialisation process required before the first time you use your Nor-840.

The battery pack may not be fully charged, so you may have to hook up the analyser to the mains via the separate power supply. Otherwise your analyser is ready for use immediately. Turn to the next page spread for more information on the batteries.

Before you start using the analyser, we recommend that you read through the *Basic Concepts* (this section, that is) of the manual to get a good understanding of the working principles of your real time analyser Nor-840.

Connecting Transducers

Microphones and other transducers are connected to the analyser's analogue sockets. These sockets are located on the *analogue socket panel.*

Connect as follows:

- Microphones are connected to the *Mic.* socket
- Accelerometers should be connected to the *Acc.* socket
- Line-drive accelerometers (accelerometers with built-in conditioning amplifier) are normally connected to the *Mic.* socket via a special adaptor
- Line level signals are fed to the Nor-840 via the *Line* input
- Intensity probe output signals are fed to the analyser via the *Intensity* probe socket.

On the Use of the Menus

The Nor-840 is menu-driven, but unlike computer operating systems with a graphical user interface (such as Microsoft™ Windows®) all menus are accessed from dedicated keys.

Inside a menu, use the FIELD CURSOR keys to navigate between the parameter fields.

You have three ways of setting a parameter:

- by using the DIAL to scroll through the available settings
- by using the PREV & NEXT keys
- by keying in a numerical value.

Even parameters that you do not normally associate with numerals have been assigned a value to enable quick setting.

Example: If you key in 1 the *Input source* menu the input source will be set to *Line* input. If you key in 2, it will be set to *Mic* etc.

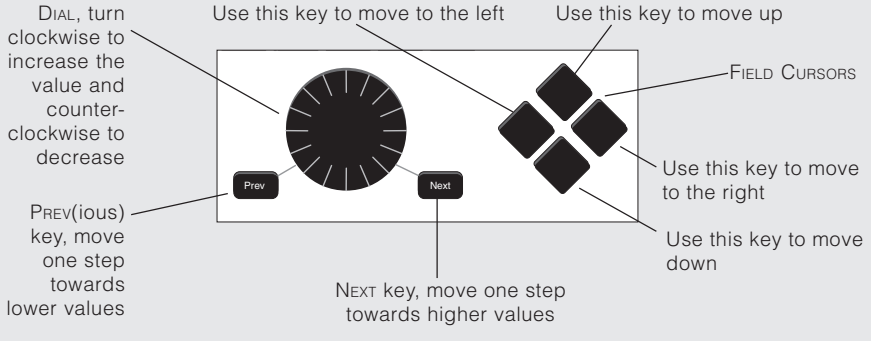
Note that when you enter numerical values the analyser's keys are in alphanumeric mode. This means that values like 10^{-6} conveniently can be entered as 10e–6.

The display can be set up to show one or two graphs – with or without essential setup information.

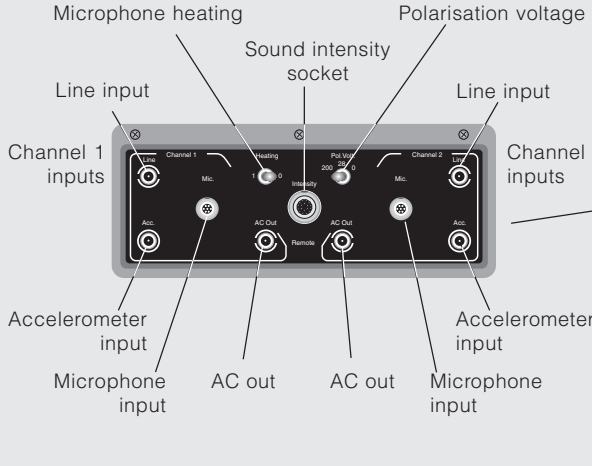
A numerical listing of the acquired data may be substituted for any of the graphs to provide a tabulation of the results.

The two display halves – referred to as the *upper* and *lower windows*, respectively – are completely independent of each other and may be set to show two unrelated functions or measurements, if required. However, the two graphs must have been made in the same measurement mode (e.g. FFT).

When Using the Menus, Several Keys Apply, in Addition to the Numerical Keypad!

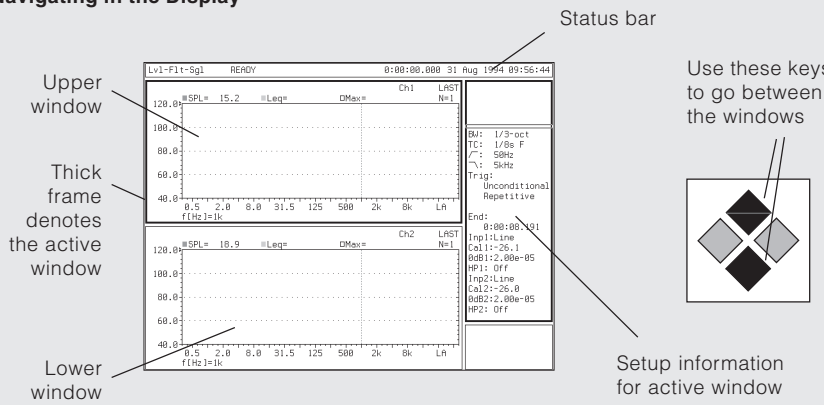

DIAL, turn clockwise to increase the value and counter-clockwise to decrease
Use this key to move to the left
Use this key to move up
FIELD CURSORS
Use this key to move to the right
Use this key to move down
Prev
Next
PREV(ious) key, move one step towards lower values
NEXT key, move one step towards higher values

Connecting Microphones & Transducers to the Analyser



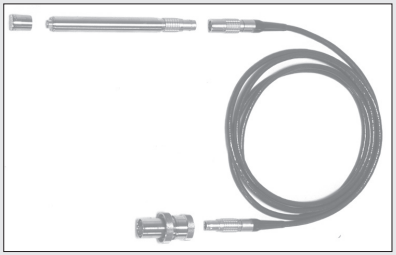
The analyser can also be delivered with B&K type connectors

Navigating in the Display


Status bar
Upper window
Thick frame denotes the active window
Lower window
Use these keys to go between the windows
Setup information for active window

The active window will respond to your cursor movements etc., the inactive will not (unless cursor alignment is activated)

Assembling the Microphone System



Always screw the microphone cartridge onto the preamplifier *before* you connect the preamplifier to the analyser.

Screw only finger tight!

To force the hard disk to stop turning (to reduce the noise emission from the Nor-840) press the function key F12.

To turn off the display backlight (to save battery power) press 2ND F12.

Repeat to turn back on (toggle function)

Which extensions are installed in my analyser?

Press F12 to produce the below picture.

```
Version information

Serial no.: 18682
Version   : 2.42
DSP1 RAM  : 32K
FLASH 1   : 2MB
FLASH 2   : not present
CPU       : 486
FPU       : present
Options   :
Dual channel, DOS, Noise generator,
Reverberation, Narrowband, Intensity,
MLS, FFT, Demo

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```

Batteries

If the battery pack is charged, you won't need external power. The standard battery pack permits two hours of use before re-charging is needed, but battery packs with higher capacity are also available. The battery pack is located inside the detachable battery box.

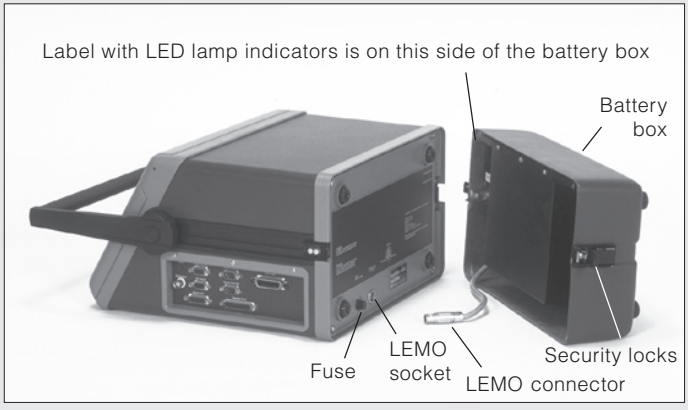
Your Nor-840 may be used with or without external power. External power is fed to the unit either directly or via the battery box. In the latter case batteries will be recharged while the analyser is available for normal use.

The LEMO connector of the power supply fits in the socket of the battery box as well as in the socket on the rear panel of the analyser – and nowhere else! The battery pack contains a battery charger and the batteries will therefore be charged as long as power is received from the power supply.

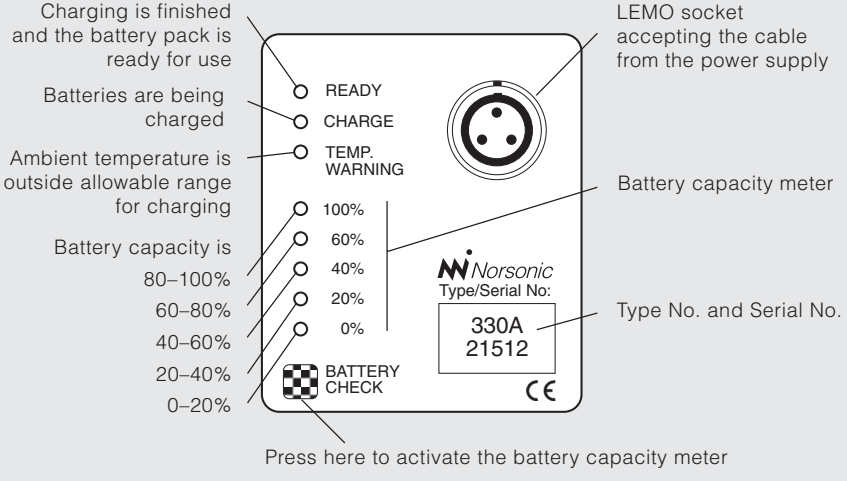
The battery box is equipped with eight LED lamps indicating the status of the batteries.

The power supply of the Nor-840…

The Battery Box is Detachable



The label with the LED lamp indicators…



Charging the Batteries

Once you plug the cable from the power supply into the battery pack, charging will commence – irrespective of whether the pack is connected to the Nor-840 or not.

The power supply supplies enough power to let you run the instrument while the batteries are being charged.

The yellow LED lamp denoted CHARGE will illuminate and stay lit as long as the charging goes on.

When the batteries become fully charged the yellow LED lamp will extinguish and the green READY lamp will illuminate instead.

If the temperature should fall outside the allowable range for charging, any ongoing charging will immediately be suspended and the TEMP. WARNING lamp will illuminate *in lieu* of the green READY or yellow CHARGE lamps. This is a safety measure to protect the batteries from damage.

It is not likely that you will encounter this under normal conditions, but it may occur if the charger is left on (i.e. connected) for a prolonged time after the battery has been fully charged.

Top-off charging will go on for approximately 10 hours after the batteries are reported to be fully charged. This will further increase the amount of energy in the batteries. During top-off charging the internal temperature in the batteries will increase and sometimes exceed the maximum permitted level.

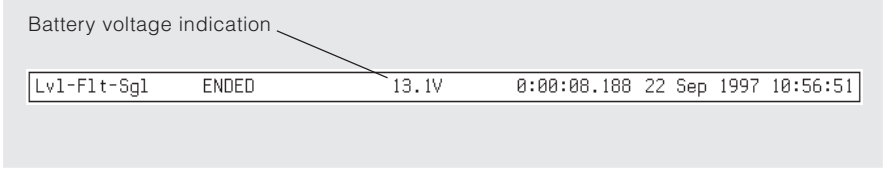
Typically this will occur if you initiate charging repeatedly, without allowing the batteries to cool in-between. Therefore you should leave the power supply connected to the mains even when not using the Nor-840, or you should only connect it to mains when charging is really needed.

The Battery Capacity Indicator

The capacity indicator (or fuel gauge) provides an indication of the remaining energy in the battery pack. When the Nor-840 is off and no charging takes place the remaining energy level is checked by pressing the BATTERY CHECK button.

When the battery pack is being charged or the analyser is operating, the capacity indicator remains illuminated.

Checking the battery capacity could be worthwhile before going on in-situ measurement sessions and is therefore strongly recommended.



Note: When the battery voltage drops below 11V, the battery voltage will appear in reverse video on the screen. A message warning you will also appear. If you are running an MS-DOS program when this happens, the LCD screen will start to flash to inform you about the battery condition. No other warning will be given!

Once the voltage drops below 10.5V a message will inform you that the analyser is about to shut itself down. Any ongoing activities will be aborted and autosave initiated as soon as the voltage gets too low to run the analyser. The Nor-840 does not discriminate between batteries and mains. Any power supply voltage drop will be considered as a battery failure.

Tip: The battery capacity indicator has been adjusted to show the battery energy level as accurately as possible. However, the battery capacity will actually increase for batteries used correctly (i.e. not exposed to repeated short-duration charging – see the article on this page for more on this), but the capacity indicator will not automatically be made aware of the increase in capacity.

To teach the capacity indicator the relationship between the actual energy level and maximum capacity, we recommend that you – once a month or once every second month – charge the batteries completely, switch on the analyser and leave it on running solely on batteries until the 0% indicator starts blinking. Then you should connect the power supply to the battery pack and recharge the batteries. The capacity indicator will now show the actual energy levels.

Your Nor-840 is a menu-driven instrument. Each menu is activated by a dedicated key on the front panel. There are no submenus.

A menu contains several parameter fields to control the parameter settings and sometimes also information fields.

You have three ways to set a parameter:

Method 1: The **PREV** and **NEXT** keys are used to scroll through the legal range of settings in the best resolution available. Use **PREV** to move towards lower values and **NEXT** towards higher. When you reach the end of a range the value will overscroll.

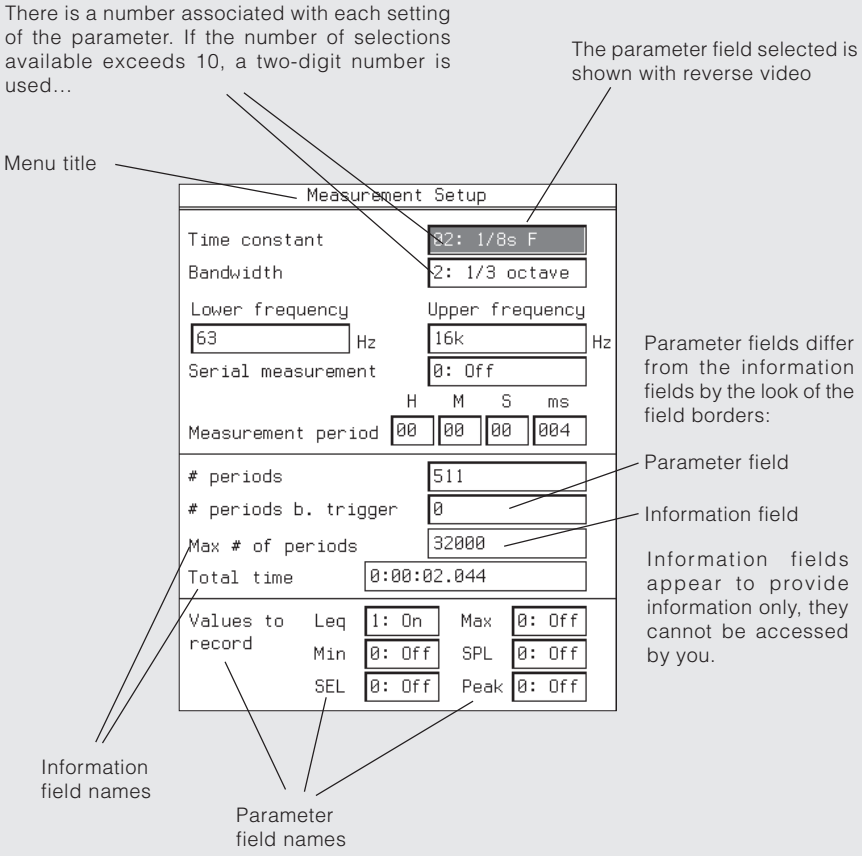
Method 2: The **DIAL** is used to scroll through the legal range of settings in a much faster way than will be the case when using **NEXT** and **PREV**. Turn clockwise to move towards higher values and counter-clockwise to move towards lower values. When you reach the end of a range the value will overscroll.

Method 3: For purely numerical values, the value to enter will be the parameter setting itself. Parameter fields with a limited number of settings also have a number associated with each setting. If you key in this number, the parameter will be set to the corresponding state immediately.

Some parameters have a two-digit number of choices. Values below 10 should then be keyed in as 0X in which $0 \leq X \leq 9$.

Note that the keyboard is in alphanumeric mode when accepting numerical inputs. This means that numbers like 10^{-6} may be conveniently entered by pressing 10e–6, i.e. using the letter e as in scientific computer notation.

A Menu Example (to enter a menu press the corresponding key)



Menus are context sensitive, i.e. some parameter fields appear when applicable only.

Navigating Through the Menu and Inside a Parameter Field

Navigating Through the Menu

Use ↑ to move to the nearest possible field above the current field. If none is available, current position is maintained.

Use ↓ to move to the nearest possible field below the current field. If none is available, current position is maintained.

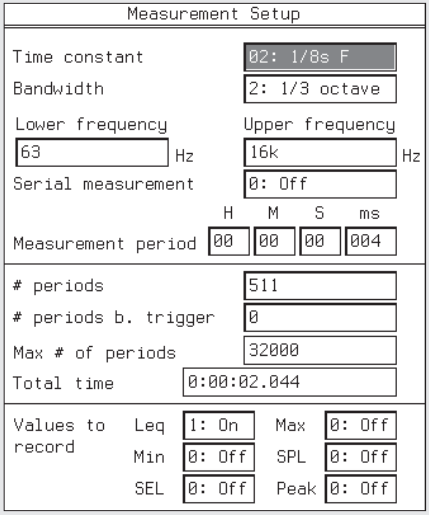
Use ← to move to the previous field in the horizontal plane. If none is available, it will move one field up. If at the beginning of the menu, overscroll (wraparound) will take place.

Use → to move to the next field in the horizontal plane. If none is available, current position is maintained. If at the end of the menu, overscroll (wraparound) will take place.

Use HOME to move to the beginning of the menu. However, not inside a list field.

Use ESC to exit a menu ignoring all changes.

Use ENTER to leave a menu, putting changes made into effect.



Inside a Parameter Field

Use INSERT to edit a string without overwriting the contents. When INSERT is operated the cursor will appear at the end of the current field.

Inside a parameter field whilst editing in the string field…

Use alphanumeric characters ASCII 32–127 to enter the contents.

Use INSERT to toggle between insert and overwrite.

Use ← or PREV to move cursor left

Use → or NEXT to move cursor right

Use DIAL to move the cursor in either direction.

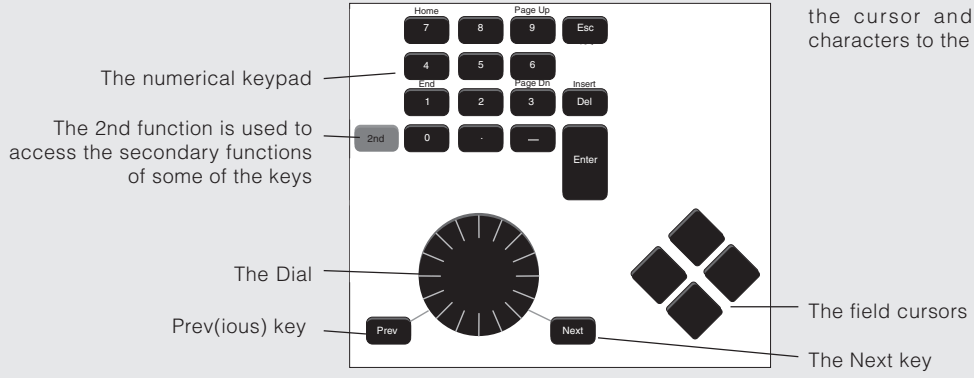
Use HOME to go to the start of the string and END to go to end of the string.

Use ESC to exit string field ignoring (undoing) all changes.

Use ENTER to leave string field, putting changes into effect.

Use DEL to delete characters to the right of the cursor and BKSP (F12) to delete characters to the left of the cursor.

The keys for these tasks…



Getting On-line Help

The Nor-840 comes with an on-line help system providing explanation of the function of an activated menu and legal ranges for the menu's parameter settings.

To display the on-line help text for a given menu, produce the menu by pressing the corresponding key and then press the Help key.

In addition the help system can be used to provide you with warnings whenever you are about to jeopardise measured or stored data and whenever you make an attempt to perform something illegal in the current state or mode.

You can set the amount of help text and warnings supplied in accordance with your demands. To set the help level, press 2ND HELP and then select the help level required.

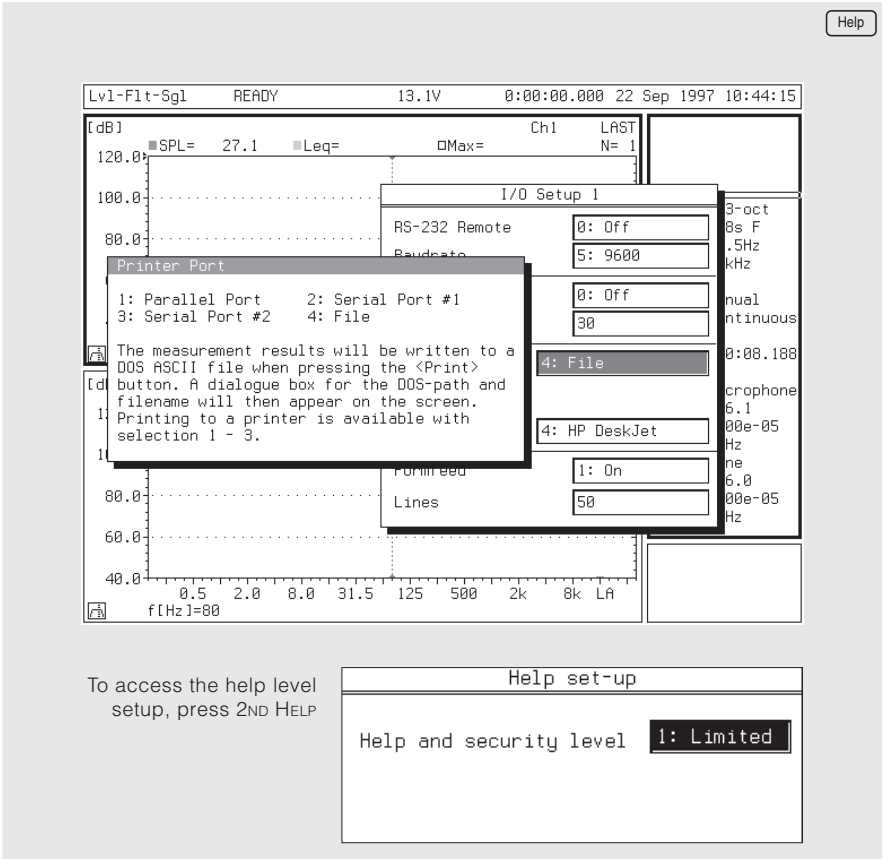
You have three levels to choose from; *Limited, Normal* and *Extended*.

Irrespective of the help level setting your Nor-840 will always provide system error information and general information such as"Calculating","Printing","Averaging"etc.

Limited will then provide no information further to what is always provided irrespective of the help level setting. This applies even to situations where jeopardising commands are about to be executed.

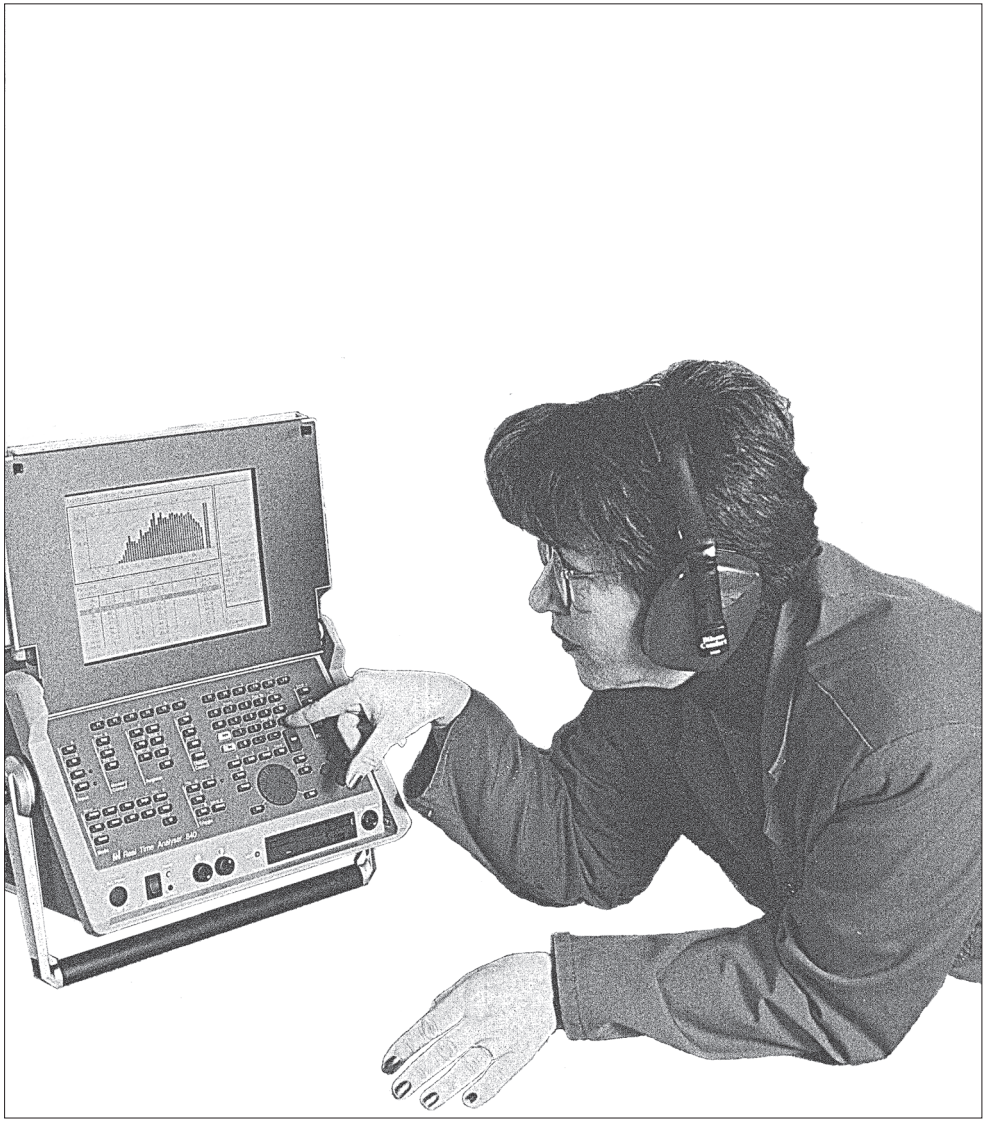
Normal will in addition provide information like"Overwriting existing file?","No file selected", "Cancel all changes?", "Measurement parameters have been changed" etc. wherever applicable.

Extended will in addition provide messages like"Number entered contains illegal characters" etc. wherever applicable.



To access the help level setup, press 2ND HELP

Tip: Start by setting the help level to *Extended* and then proceed to *Normal* as you get more skilled and experienced. Leave the *Limited* setting to true expert users only.



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Making Level Measurements

The level mode is the basic measurement mode in the Nor-840. With this mode you can make measurements of the level vs. frequency in octave bands or fractional octave bands. Your measurement can yield a single spectrum summing up what took place during the measurement or it can consist of a series of consecutive measurements all of the same duration. In the latter case you will be able to produce a time profile of your measurement by looking into a single frequency band at the time to see how the level of this frequency band changes as a function of time. Using only one spectrum to document what went on is referred to as a single spectrum measurement, while the other method is referred to as multispectrum measurement. More information about this can be found in the article *Single Spectrum vs. Multispectrum* on the next page.

Whenever you are going to set up the instrument for a level vs. frequency measurement you need to set the following:

- Input source selector
- Full scale setting
- Calibration
- Measurement mode
- Measurement parameters
- Trigger conditions

Input source selection is used to activate the input terminal to which the transducer is connected. Whenever using microphones, set the input source selection to Microphone.

Full scale setting is used to set the input amplifier gain so that you get the optimum use of the analyser's dynamic range.

Calibration is needed to ascertain that the values measured are accurate enough to serve their purpose.

Measurement mode selection refers to the selection between single spectrum and multispectrum measurements, given that you already did select Level.

Measurement parameter settings are used to define the time constant, the frequency range, which functions to measure and the measurement duration. For multispectrum measurements the number of periods (the number of consecutive measurements) should also be defined.

Trigger condition settings are used to define the start conditions of the measurement. Should it start when hitting the START key? Or, should it start when the level exceeds a given threshold? At a given moment in time? Should there be any delay from trigger until the measurement actually starts? Altogether you have nine trigger condition alternatives at your disposal.

To begin measuring press the START key. The data acquisition will then start as soon as the trigger condition is met and go on until the preset measurement time expires – if left uninterrupted.

Pause. You may pause the instrument at any instant during the measurement. Press CONT to resume.

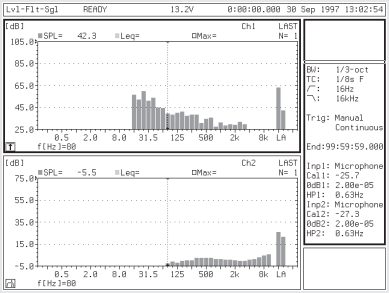
Prolong. You may prolong a measurement if the initial duration proved insufficient.

Merge. You may merge the two channels of a dual channel measurement into a single channel measurement. This feature is often used to save time when making measurements involving spatial averaging.

Averaging. In order to reduce the influence of extraneous noise, measurements are often averaged together. We have even included an *undo average* function in case you make an average too much.

Serial analysis. A multitude of real time analysers on the market have – for some strange reason unknown to us –been made without the ability to make serial frequency analysis. From time to time, you will probably encounter situations where an excitation signal is needed, but the source turns out be incapable of generating a sufficiently high output level. However, if you bandlimit the output signal you will normally be able to obtain an output with a higher level. This can only be utilised if your analyser can do serial analysis, i.e. one frequency band at the time. The Nor-840 can do serial analysis in manual as well as in automated (scanning) mode.

Starts from Level Mode – Always!



When you switch on the instrument, it will always come up in Level mode. To switch to Level mode from any other mode press the LEVEL key followed by SINGLE or MULTI.

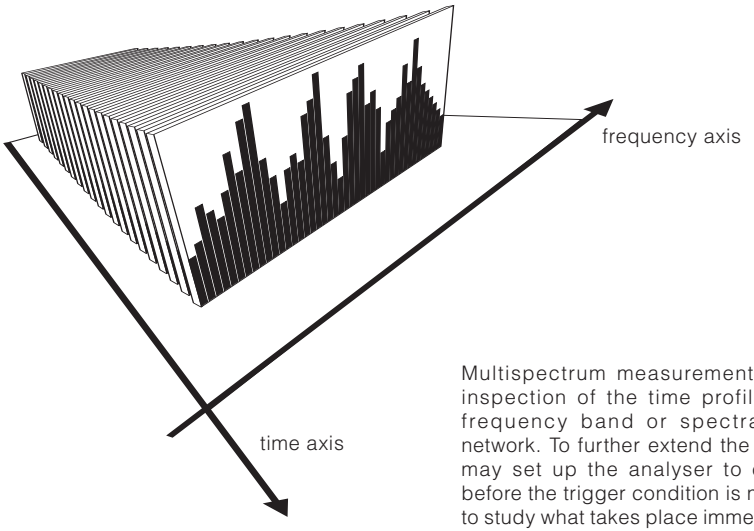
All measurements made with the Nor-840 are either single spectrum or multispectrum measurements, irrespective of the measurement mode in which the measurement was made.

A multispectrum measurement is a recording of successive measurements, all having the same duration.

The successive measurements are referred to as *periods*, and the duration of each period is referred to as the *period length*. The period length can be set to anything between 4 msec and 100 hours. If more than one function is to be measured (e.g. L_{EQ} and SPL_{MAX} simultaneously) the lower limit is 10 msec.

At any rate, the maximum number of periods in a single multispectrum measurement is limited by the amount of free memory available in combination with the number of functions to be measured simultaneously.

Single spectrum measurements are multispectrum measurements with the number of periods available fixed to one. The single spectrum mode appears for convenience only. The primary application of the single spectrum mode is general level or intensity analysis, while multispectrum mode is used principally to obtain time profile information such as capturing sound decays for reverberation time analysis.



Multispectrum measurements permit the inspection of the time profile of a given frequency band or spectral weighting network. To further extend the overview you may set up the analyser to capture data before the trigger condition is met to be able to study what takes place immediately before the trigger condition is fulfilled.

Upon a successful multispectrum measurement you may inspect the time profile for a given frequency band or spectral weighting function as well as inspect the entire amount of acquired data as a three-dimensional plot with selectable viewing angle.

The two modes – the single spectrum mode and the multispectrum mode are completely independent of each other and have separate measurement setups. This makes building acoustics measurements with the Nor-840 particularly easy, as the setups for sound reduction measurements (single spectrum) and sound decay captures (multispectrum) may be defined in beforehand and will continue to be kept apart so that you won't have to define them again on-site.

However, the above segregation does not include global settings such as input selection, gain and calibration settings.

To go from the single spectrum mode to the multispectrum mode, press MULTI and to go the other way press SINGLE.

Since the two modes are independent of each other, going between the modes will have no effect on acquired, but not yet saved data. Each of them has its own part of the memory. However, they share a common memory pool, if one of them is particularly large, this may affect the amount of memory available to the other mode.

Note that the instrument will not accept that you switch between the two during a measurement.

You cannot measure in more than one mode at the time.

Selecting Input Source for Level Mode Measurements

The Nor-840 has four different types of signal input sockets. These are accessed via the input menu. Press the INPUT key to produce this menu.

Only one input can be selected at the time per channel, but the two channels need not be set to the same type of input source.

One of the channels may be set to *Off*, but both channels cannot be set to *Off* at the same time.

If you set one channel to *Off* in the input source menu, the corresponding part of the calibration menu will be blank.

Observe that if you select sound intensity for one of the channels, the other channel will be set to the same mode automatically, since sound intensity is a two-channel measurement technique. If you then change the setting of one of the channels, the other will go back to the setting it had at the time intensity was selected.

A lowpass filter will be activated when *3: Charge* is selected.

The menu contains a highpass filter for each channel. If you activate the highpass filter, it will affect the setting of the lowest frequency band to be measured.

This setting is found in the measurement setup menu.

The new setting will be the first centre frequency above that of the highpass setting provided the initial setting was lower than or equal to the highpass filter is set to. If it was set to a higher value initially, it will remain unaffected.

Note that although the input selection is a global setting (affecting the entire instrument and all modes), only the current mode (single spectrum or multi-spectrum) will be affected by the forced change of lower end frequency range setting.

The reason why the other modes are not affected, is that otherwise measured data not stored would be deleted to avoid inconsistencies.

However, this means that if you make a measurement in any other mode without modifying the lower end setting of the frequency range, the frequency bands in the vicinity of the highpass filter cutoff frequency will be biased due to the presence of the highpass filter.

In the current mode, you may change the setting of the lowest frequency band to be measured, to the value of your liking *after* the highpass filter has been activated.

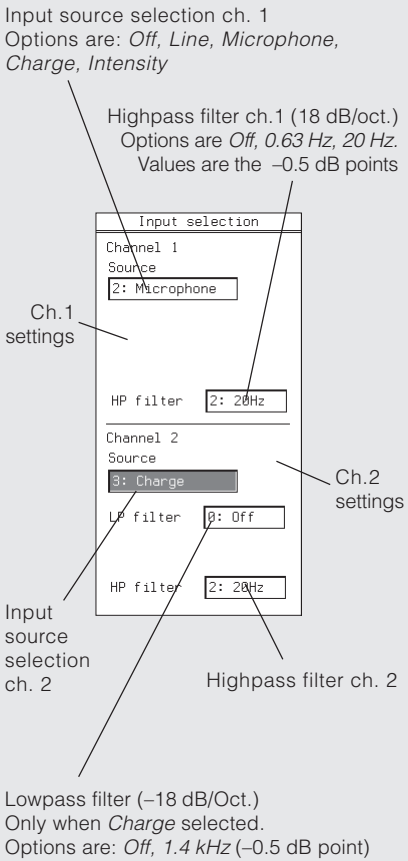
The change is made inside the measurement setup menu for the current mode.

Should you again modify the highpass filter setting and set it to a higher value (in Hz), the same will take place again provided the lower end setting is lower or equal to the highpass filter cutoff frequency. Otherwise it will be left unaffected.

If you decide to modify the lower end keep in mind that the above biasing of the lower frequency bands will apply to the current mode as well.

The highpass filter cutoff frequency stated is the –0.5 dB frequency point (not the usual –3 dB frequency point).

The Menu for this Task Input



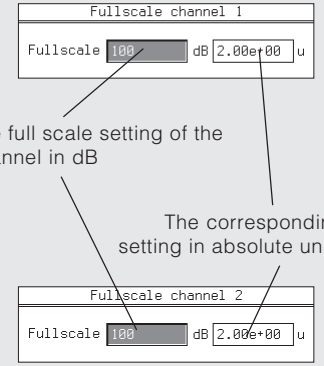
If a channel is set to *Off* in this menu, the corresponding part of the Calibration menu will appear blank.

The Menu for this Task

Gain1

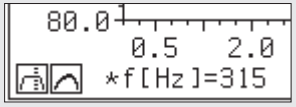
Gain2

There is one menu per channel


Fullscale channel 1
Fullscale 100 dB 2.00e+00 u
The full scale setting of the channel in dB
The corresponding setting in absolute units
Fullscale channel 2
Fullscale 100 dB 2.00e+00 u

Do not confuse the full scale setting and the full display setting! The former defines the input amplifier gain and hence the overload margin for a given signal level, while the latter is used to set the display to make the measured signal fit within the setting of the axis.

If you set the full scale setting to one of the three highest settings, a small * will warn you that the analyser is prone to overload...



The available full scale setting of a measurement is given as a combination of the input amplifier gain setting and the calibration setting.

The gain is set separately for each channel and is adjustable in steps of 5 dB.

Press GAIN1 (GAIN2) to enter the menu.

The full scale setting controls the setting of the input amplifiers. However, it has no influence on the vertical scale in the display. The two extremes (top and bottom) of the vertical scale is controlled by the *top scale value*, which has its own dedicated key called the Y-MAX key. The top scale setting is purely a display function having nothing to do with the input amplifier whatsoever.

If you are uncertain about which gain setting to use, there is an *autoranging* feature available.

The autoranging is separate for each channel and is activated by pressing 2ND GAIN1 (GAIN2).

Once activated the instrument will set the full scale to maximum and wait for three seconds to let the analogue circuitry settle. The analyser will then measure the SPL twenty times at 100 msec intervals. Five decibels will be added to the highest SPL detected and the result will be rounded off upwards to the nearest standard gain setting.

Note that if you set the full scale setting to a very high value, you face the risk of having severe overload in the transducers without seeing any trace of it in the analyser – simply because the analyser input isn't overloaded. For example; a ½" microphone with a sensitivity of 50mV/Pa will distort severely (more than 3% total harmonic distortion) when exposed to levels above 135 dB(A). To warn you about this the three highest settings provide an asterisk (*) in the display.

Calibrating for Level Mode Measurements

The Nor-840 is calibrated by means of a sound or vibration calibrator and the calibration menu.

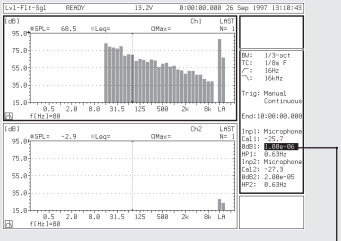
Although you may calibrate by just keying in the sensitivity, we always recommend that you use a calibrator. This is the only way to ensure proper operation of the entire measuring chain including the transducer(s).

If you are going to use the instrument for vibration measurement it may be convenient to change the 0 dB level to obtain dB readings easy to compare with other vibration measurements. Check with relevant Standards and conventions to find suitable or commonly used 0 dB levels.

Note that if you set the 0 dB level to a value different from 2×10^{-5}, the selected setting will appear in reverse video on the screen as shown in the lower Fig. to the right.

Half of this menu may occasionally appear blank. This will take place whenever the corresponding input channel has been set to *Off*.

If you Change the 0 dB Reference Level



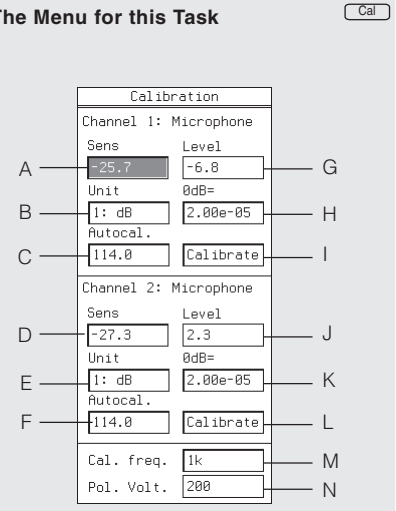
...an indication is provided

The Tool for this Task

Norsonic offers three different sound calibrators (all available separately) to cover the requirements of the IEC 942 class 0L, class 1 and class 2. The model shown here is our class 1 calibrator, the Nor-1251. The calibrator is shown at approximately 50% of actual size.

Note th adaptor designed to permit use of 1" as well as ½" microphone cartridges.

The Menu for this Task

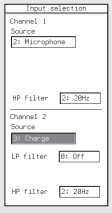


A Sensitivity of channel 1
B Ch. 1 units (dB or engineering units)
C Auto-calibration level of channel 1
D Sensitivity of channel 2
E Ch. 2 units (dB or engineering units)
F Auto-calibration level of channel 2
G Level of ch. 1 measured with the selected time constant
H 0 dB level of ch. 1
I Initiate auto-calibration of ch. 1
J Level of ch. 2 measured with the selected time constant
K 0 dB level of ch. 2
L Initiate auto-calibration of ch. 2
M Calibrator frequency
N Polarisation voltage setting

Calibration Using a Sound Calibrator

❶

Insert microphone into calibrator

❷ 

Press INPUT and select input source

❸

Adjust the full scale setting (GAIN1/GAIN2)

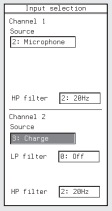
❹

Press CAL and set the calibrator frequency

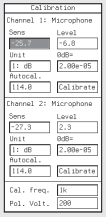
❺

Set the sensitivity of the microphone used until the sound pressure level is indicated correctly

Calibration, Setting the Sensitivity

❶ 

Press INPUT and select input source

❷ 

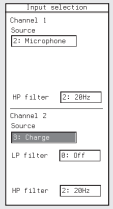
Press CAL and set the sensitivity of the microphone used

This method is in general not recommended since it does not take into account whether the microphone actually works as it is supposed to do. Bearing in mind that the microphone itself is the most vulnerable part of the measurement chain, this method obviously suffers from severe shortcomings.

Auto-calibration, Using a Sound Calibrator

❶

Insert microphone into calibrator

❷ 

Press INPUT and select input source

❸

Adjust the full scale setting (GAIN1/GAIN2)

❹

Press CAL and set the calibrator frequency

❺

Set the calibrator's output level

❻

Select *Calibrate* and press ENTER to commence auto calibration

The measurement setup menu is used to set up essential measurement parameters.

The menu comes in two flavours; one to cover the setup of single spectrum measurements and another to cover that of the multispectrum level measurement.

To access the menu press the M.SETUP key.

If you alter the high pass filter setting of the input selection menu, this will cause the lower setting on the frequency range to be changed if the initial setting was lower than that of the new high pass filter setting.

If it was set to a higher value initially, it will remain unaffected.

You may change the lower end setting of the frequency range in the measurement setup menu after having set the high pass filter in the input menu. However, if you set the frequency range to include frequency bands below the high pass filter setting, the readings of these frequency bands will be biased because of the high pass filter.

Note that changing the high pass filter setting affects the measurement setup menu of the selected mode (single spectrum or multispectrum) only. Other modes will not be affected (and may hence yield biased data) to avoid inconsistencies.

Multispectrum measurements consist of a series of consecutive measurements referred to as *periods*. All periods will have the same duration (specified in the measurement setup menu) and there is no loss of data in-between the periods.

You may set up your Nor-840 to capture the time profile of what goes on immediately before the trigger condition is met. This is done by specifying that a certain number of the multispectrum periods shall apply to the situation before trigger.

Although this may sound like a violation of the causality requirement a brief look into how it is done will reveal that causality is indeed maintained.

Having set up for recording of periods before trigger, the instrument will start to acquire data once you press the START key. However, the acquired data are put in a circular buffer having a length exactly matching the number of periods before trigger.

When the buffer gets full, the oldest data are overwritten. In this way the buffer will always contain the latest periods acquired.

Once the trigger condition is met, the contents of the circular buffer will be the time profile required. This is then retained while the measurement goes on as usual.

Note that to make this work properly the time elapsed between the START key is pressed and the trigger condition met must be equal to or grater than the product:

#periods before trigger × *period duration*

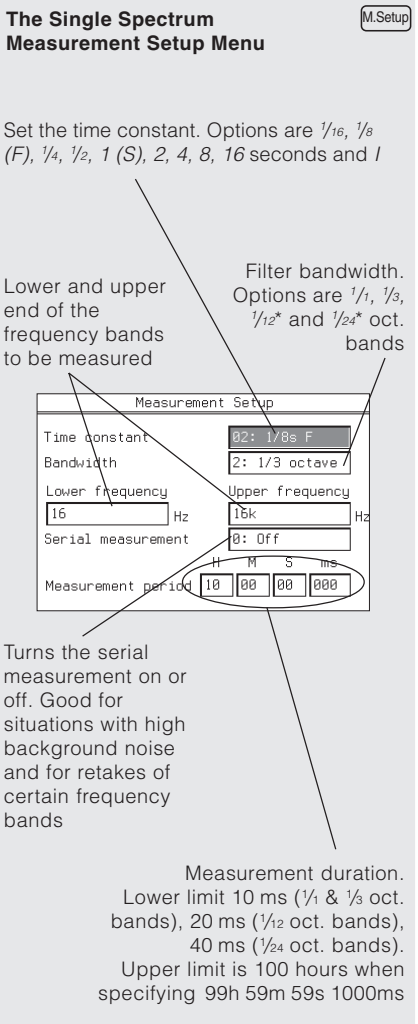
which expresses the total time to be spent on the pretrigger time profile recording.

If the trigger moment comes before this, only the periods recorded so far will be shown while the rest of the periods will be set to zero.

Note that in order to make this work, the trigger condition cannot be set to manual.

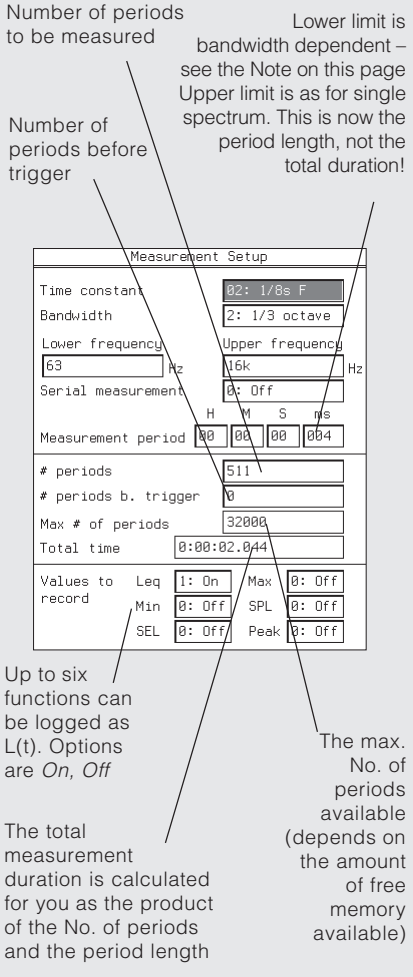
For obvious reasons, the above situation applies to multispectrum measurements only. The single spectrum mode should be considered as a collapsed multispectrum measurement where the number of spectra is one.

The Single Spectrum Measurement Setup Menu


M.Setup
Set the time constant. Options are 1/16, 1/8 (F), 1/4, 1/2, 1 (S), 2, 4, 8, 16 seconds and I
Lower and upper end of the frequency bands to be measured
Filter bandwidth. Options are 1/1, 1/3, 1/12* and 1/24* oct. bands
Measurement Setup
Time constant
02: 1/8s F
Bandwidth
2: 1/3 octave
Lower frequency
16
Hz
Upper frequency
16k
Hz
Serial measurement
0: Off
H M S ms
Measurement period
10 00 00 000
Turns the serial measurement on or off. Good for situations with high background noise and for retakes of certain frequency bands
Measurement duration. Lower limit 10 ms (1/1 & 1/3 oct. bands), 20 ms (1/12 oct. bands), 40 ms (1/24 oct. bands). Upper limit is 100 hours when specifying 99h 59m 59s 1000ms

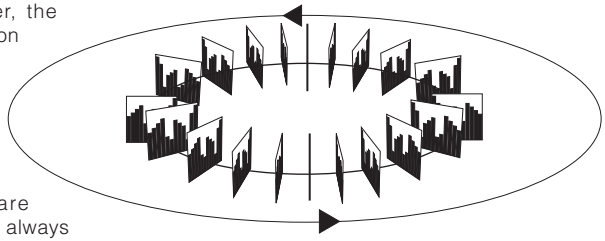
* Optional, must be ordered separately

The Multispectrum Measurement Setup Menu is an just extension of the single spectrum setup menu…

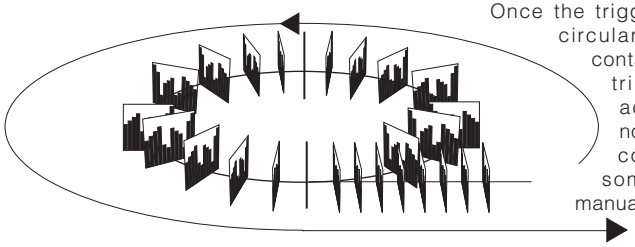


In multispectrum mode the Nor-840 can be set up to record a number of spectra before trigger…

When you have set up the Nor-840 to record a part of the periods before trigger, the acquisition of data starts upon pressing the START key. The data are stored in a circular buffer whose length exactly matches the number of periods to be recorded before trigger. When the buffer gets full, the oldest data are overwritten. Thus the buffer will always contain the latest periods acquired.



Once the trigger condition is met, the circular buffer will be retained containing the periods before trigger while the data acquisition will go on as normal. Note that the trigger condition must be set to something different from manual to make this feature work.



Note: The lower limit for the period length i multispectrum mode is bandwidth dependent; 4 ms (1/1 & 1/3 oct. bands), 10 ms (1/12 oct. bands) and 20 ms (1/24 oct. bands). If more than one function is set active, the lower limits are 10 ms (1/1 & 1/3 oct. bands), 20 ms (1/12 oct. bands) and 40 ms (1/24 oct. bands).

Note: The time constant 1/16 s is too short to produce meaningful results when measuring in 1/24 oct. bands, this combination of settings has therefore been disabled and cannot be selected. Switching to a less narrow analysis bandwidth or a longer time constant will restore valid combinations.

Tip: In multispectrum mode, the number of periods available depends on the amount of free memory available. However, if you can do with fewer functions logged simultaneously, or with a more narrow range of frequency bands, the free memory available can be spent on what you really need. The number of periods available will be doubled if the number of functions logged is halved. Likewise it will be doubled if the frequency range is halved.

Therefore, we recommend that you consider your needs before you set up the analyser. Is the time domain resolution you have chosen appropriate – or is it overkill, drowning you in data? If you can get away with, for instance, half the time resolution (i.e. doubling the period length, you will be able to cover the same total duration with only half the number of periods (so maybe you could keep that many functions activated after all?).

Trigger Conditions in Level Mode

There are separate trigger setups for each of the two modes (one for the single spectrum mode and another for the multispectrum mode). In this way the two modes are kept apart – a feature that comes in handy when making building acoustics measurements with the Nor-840.

The menu contains context sensitive elements; if you select trigger conditions related to level, extra parameter fields will appear to let you define what level in which channel and which frequency band to serve as the trigger condition.

Likewise, if you select clock as trigger condition, extra fields will appear to let you define the time of trigger.

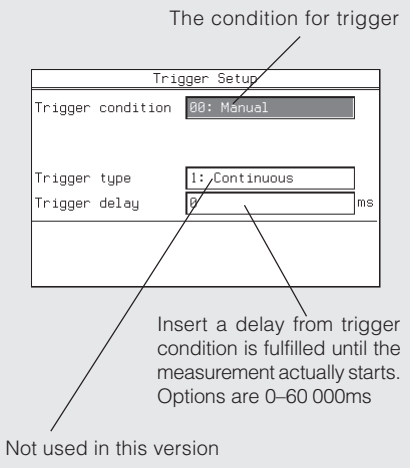
Note that if you want to record the time profile of what takes place immediately before the trigger condition is met, you may set up the analyser to do so. However, this applies to multispectrum measurements only and it is not done in the trigger setup menu, but in the (multispectrum) measurement setup menu.

The trigger condition menu is accessed by pressing 2ND START.

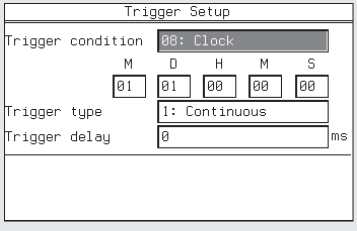
The nine trigger conditions available and how they work can be seen on the right part of this page spread.

The Menu for this Task

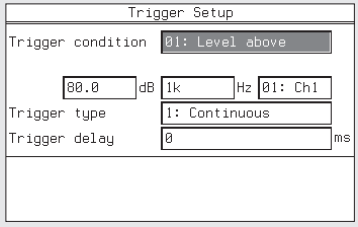
2nd Start

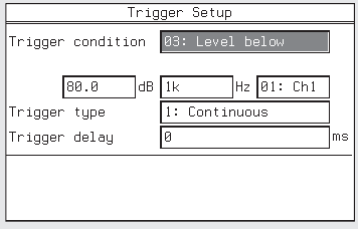


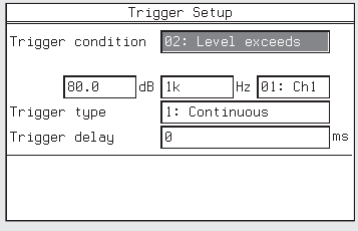
If you select Clock as trigger condition, extra parameter-fields will appear…

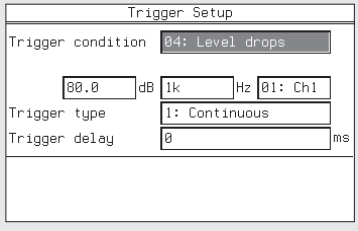


If you select level related trigger conditions, extra parameter-fields will appear…

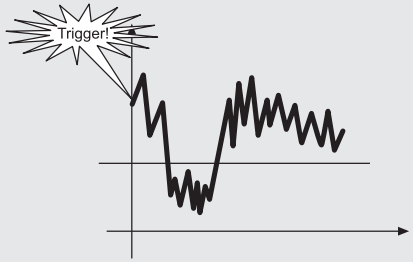






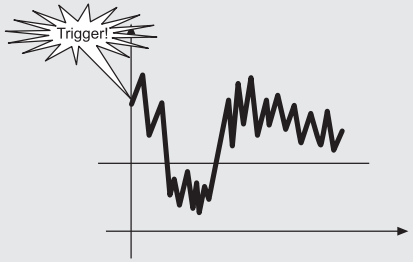


Manual as trigger condition



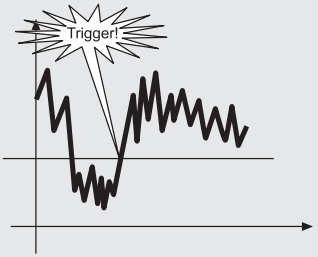
“as soon as START is pressed”

Level above… as trigger condition



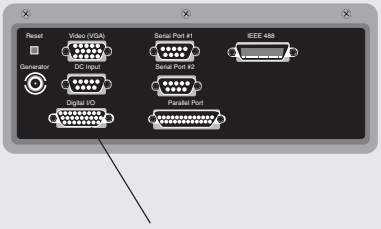
“whenever the level is above the threshold”

Level exceeds… as trigger condition



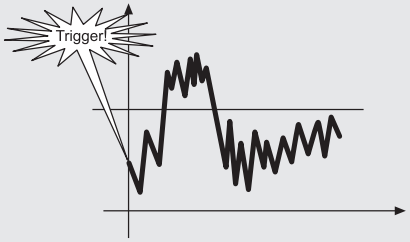
“transition is the keyword”

External… as trigger condition



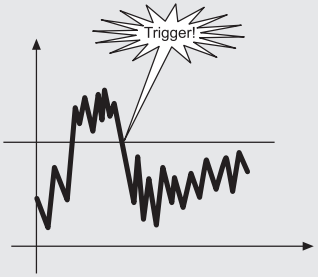
“grounding pin 23 on digital I/O will do it”

Level below… as trigger condition



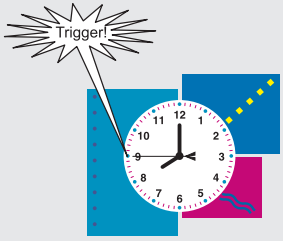
“whenever the level is below the threshold”

Level drops below… as trigger condition



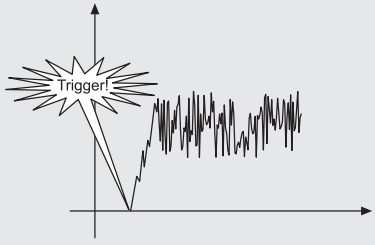
“transition is the keyword”

Clock… as trigger condition



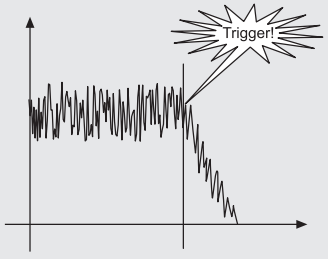
“at a specific moment in time”

Noise on… as trigger condition



“when you switch on the internal generator”

Noise off… as trigger condition



“when you switch off the internal generator”

Once the Nor-840 has been set up to your requirements, it is ready to make measurements.

To begin measuring, press the START key. The data acquisition will start as soon as the trigger condition is met. If you have set up a trigger delay, the acquisition will not begin until *a) the trigger condition has been met* and *b) the delay subsequently elapsed*.

The measurement will – if left uninterrupted – go on until the measurement end condition is met.

You may halt the measurement temporarily and then resume the measurement later. You may also stop the measurement prematurely by pressing the STOP key.

The instrument does not discriminate between pausing and termination with respect to resuming a halted measurement. Therefore, to pause the instrument just press the PAUSE or the STOP key.

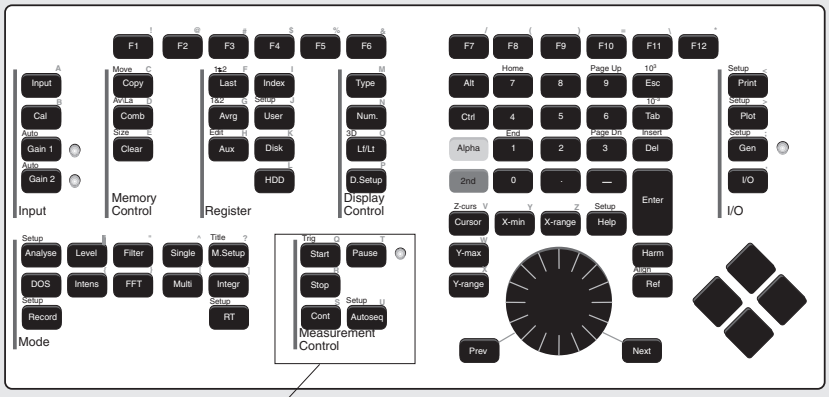
To resume measuring, press the CONT key.

If the measurement was halted prematurely (paused), pressing CONT will cause the instrument to resume the measurement and go on until the preset measurement duration expires.

On the other hand, if the CONT key is pressed after a measurement has ended successfully (i.e. the preset duration has expired) the measurement will be prolonged by a another period equal to the preset duration.

In the latter case the total duration will be the sum of the two durations (provided the measurement was not terminated prematurely during the prolongation).

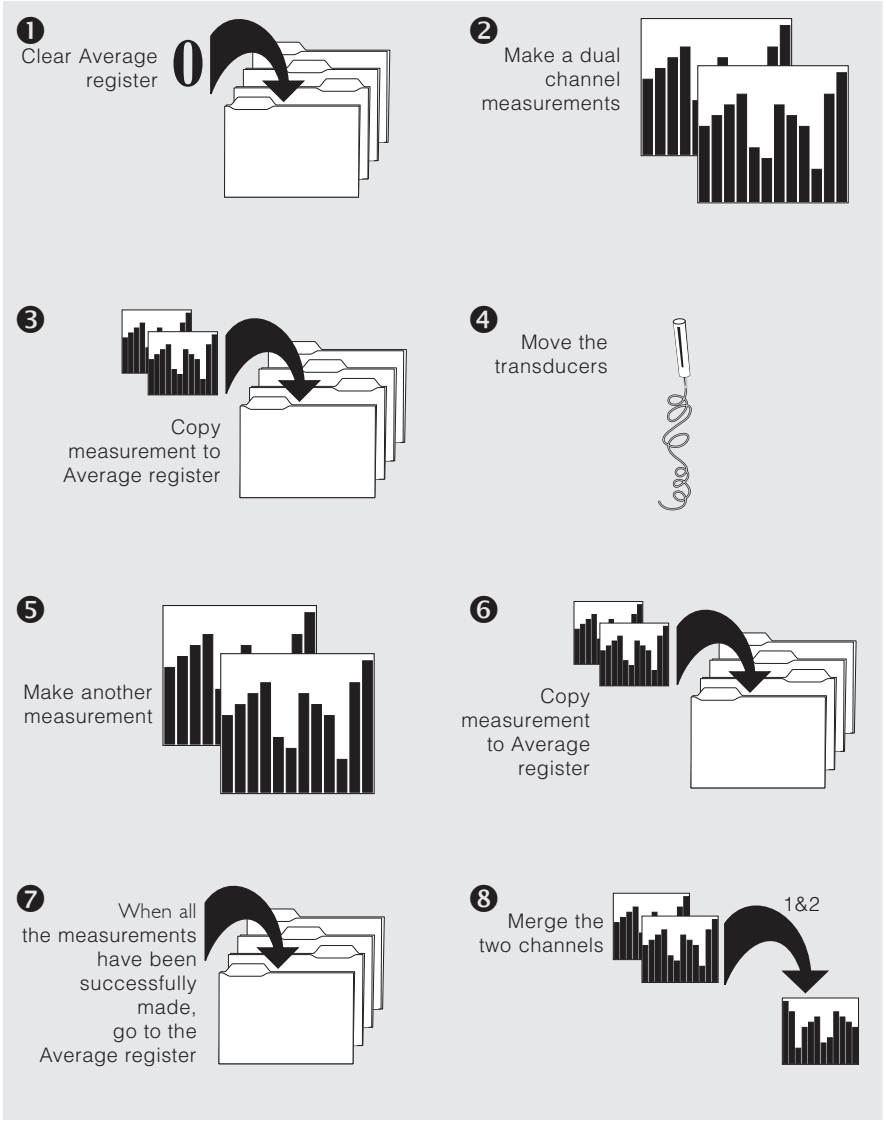
The Tools for this Task



The measurement control keys

Pressing:	Causes:
Start	the measurement to begin as soon as the trigger condition is met.
Pause	an ongoing measurement to be temporarily halted
Stop	an ongoing measurement to be terminated
Cont	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the measurement end condition is met.
Cont after the measurement end condition is met	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the measurement end condition is met. The total measurement duration will be the sum of the durations of the two measurements.

Merging Dual Channel Data to Obtain Single Channel Data



Sometimes when making measurements that require averaging, you may not need dual channel data.

Instead of switching off the second channel, try using the 1&2 function (2ND AVRG key sequence).

This function takes the data contained in channel 2 out of this channel and merges them with the contents of channel 1. The method used is energy-based averaging.

The merged data end up in channel 1. Channel 2 will be left empty.

Note that the function works with the Average register only. Do not operate the 1&2 function until all of your measurements have been successfully merged (as two channel data) in the Average register.

This is a function that can save you a lot of time. When e.g. you are measuring in rooms, you will normally need to make spatial averaging in order to obtain representative data. This will require several measurements.

If you only need single channel data anyway, why not measure in dual channel mode, merge the data and thus cut the time spent by almost half?

Serial Level Analysis

The Nor-840 is basically a parallel analyser, i.e. it measures in all the activated filter bands simultaneously.

However, there are situations where the broadband output of your noise excitation may be too low to get a signal-to–noise ration high enough for your requirements.

Enter the serial analysis function. This function turns the Nor-840 into a serial or sequential frequency analyser.

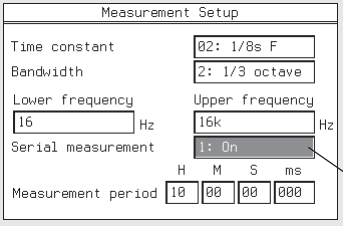
If you limit the bandwidth of the signal source output signal and combine this with serial analysis you will get a higher signal level at the expense of a more narrow signal bandwidth.

The function may also be used to polish spectra already acquired; if some of the frequency bands yield unsatisfactory results, these may be retaken without affecting the rest of the frequency band results.

The Menu for this Task

M.Setup

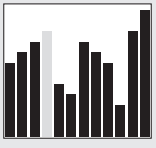
The feature is found in the Measurement setup menu and is accessible in single spectrum as well as in multispectrum mode



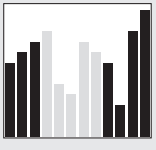
Serial measurement is activated here
Options are *0:Off; 1:On* and *2:Scan*

The serial function is a good idea to make use of when…

…you do not have sufficient signal-to–noise ratio because of limited noise source output level or too high background noise level



…you need to retake a single frequency band or a row of frequency bands



Note: When using the serial analysis function for retakes you will have to enter the measurement setup menu to activate the serial analysis function. However, you must make no other changes in this menu. Neither must you make any changes to the input, calibration nor the gain setting menus. Any changes made in these menus will cause the analyser to erase the data already acquired. If you have set the help level to Limited, you will not even be notified about the erasure of data!

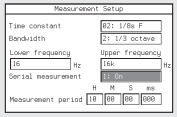
Tip: Once you select serial analysis, the noise generator will automatically switch to pink, bandpass filtered noise. If this is unsatisfactory, you may use the generator setup menu to change the setting.

When you later deactivate the serial analysis function, the noise generator will switch back to the noise type it was set to initially. Note that the other generator setup parameters will not be reset – instead the new settings will be retained.

Tip: When a serial analysis measurement is running in serial scan mode and it is stopped prematurely by you, the measurement will go on for another ten seconds or until the preset measurement duration expires – whichever comes first.

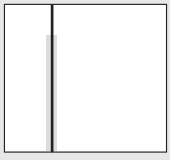
Making a Serial Measurement

❶



Set *Serial measurement* to *1:On*

❷



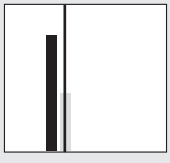
Move cursor to the frequency band to be measured

❸

Start

Press START to measure the level of the selected frequency band

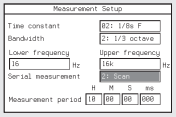
❹



Move graph cursor and repeat, if required. Use PAUSE and CONT keys as usual

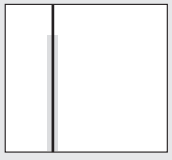
Making a Serial Scan

❶



Set *Serial measurement* to *2:Scan*

❷



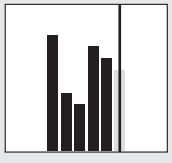
Move cursor to the lower end of the frequency band interval to be measured

❸

Start

Press START to measure the levels of the frequency band interval

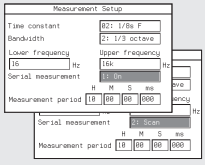
❹



Measurement will stop when the upper frequency band (as defined by the Measurement setup menu) is reached or STOP is pressed. Use PAUSE and CONT as usual

Retaking Frequency Band(s)

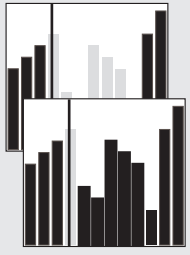
❶



Set *Serial measurement* to *1:On* or *2:Scan*

Make no other changes!

❷



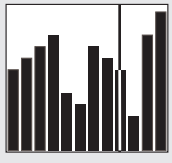
Move cursor to the lower end of the frequency band interval to be measured again

❸

Start

Press START to measure again without affecting the frequency bands to be retained

❹



Measurement will stop when the upper frequency band (as defined by the Measurement setup menu) is reached or STOP is pressed (Scan selected). If set to On, move to another frequency band and repeat

Use PAUSE and CONT as usual

Averaging a Measurement too Far

Merging measurements by averaging them together is needed in order to minimise the influence of spurious (unwanted) signals.

Now… what if you get a bit carried away and average your most recent measurement with measurements made previously before you realise that doing so is going to ruin your entire measurement sequence?

Since this is indeed not an unknown problem (most people will do this sooner or later) the Nor-840 has been furnished with an *undo average* function.

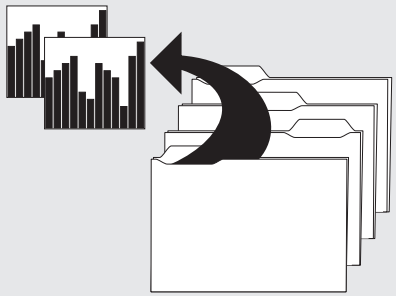
For data security reasons the functions Copy and Combine are identical when applied to the average register, so the method described applies to both functions.

The Procedure

Assume you just averaged the contents of the Average register with a measurement result that shouldn't at all be merged with the Average register contents…

You can undo this by using the Av\La (Average less Last) function.

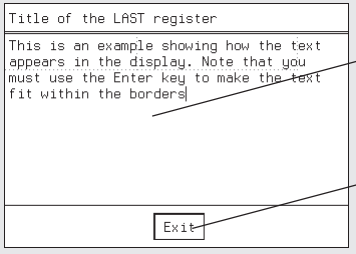
Press 2ND COMB to activate this function.



Note: There is only one level of undo, i.e. you cannot repeatedly use the undo key to undo previous mergings!

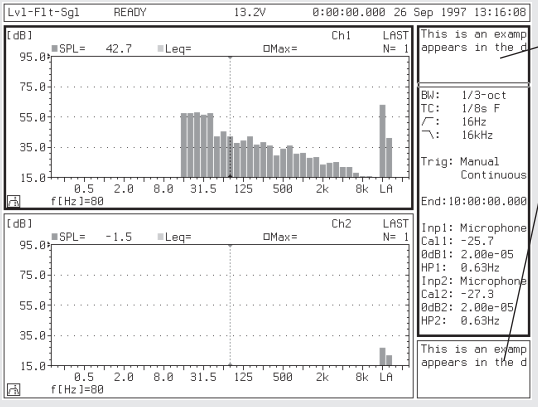
The Menu for this Task

2nd M.Setup


Title of the LAST register
This is an example showing how the text appears in the display. Note that you must use the Enter key to make the text fit within the borders
Exit

Enter the text here. The analyser will enter alpha mode upon entering this menu, so you won't have to press ALPHA first

To exit the menu, you must use the FIELD CURSOR to move to this field and then press ENTER



…and here is how the text appears in the display. Note how the text has been truncated. Only the text written inside the dotted field appears in the display when exiting the menu. The text is the same for the upper and lower window since they belong to the same measurement

You may assign a title to your measurement. This may help you keep track of what was measured and where.

The title is assigned to the contents of the current register of the current mode. If you move or copy the contents of the current register, a copy of the title follows the measured data.

If you assign a title to the Last register and let it remain unchanged, all following measurements will acquire the very same title, since the title is assigned to the contents of the Last register – whatever that may be.

We therefore recommend that you consider updating this title regularly to make it meaningful.

If you change to another measurement mode, the currently selected register of the new mode will acquire the same title as the same register of the previous mode, unless that register of the new mode had a title already and it was empty in beforehand.

To access the Title menu, press 2ND M.SETUP.

Tip: If automatic file name generation has been set to off and a title is present, the eight first characters of the title – ignoring any spaces – will be proposed as file name when storing the acquired data.

A Tour of the Level Mode Display

There are four display modes in the Nor-840 – two single function and two dual function modes.

When we say function display rather than channel display, this is because in dual function display mode, the two display halves are completely independent!

The upper display may show what you just measured, while the lower may show a graph of a stored measurement (or vice versa), or the upper display may show a graph while the lower shows a tabulation of the very same data etc.

When you select any of the two dual function displays (with or without setup information), the screen will consist of an upper and a lower part – which we may call windows – plus, whenever applicable the instrument and measurement setup information.

Only one window is active – i.e. responds to cursor movements etc. – at the time. The active window has a thick frame surrounding it.

Use the vertical pointing arrows of the ❖ (FIELD SELECTOR) keys to go between the windows.

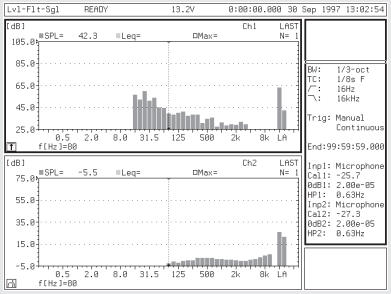
In each window you may set such display parameters as horizontal and vertical graduation, the vertical top scale and the horizontal datum (the lower end of the scale) as well as move the graph cursor.

Observe that the alterations possible in the display have no effect on the measured or recalled data as such, only the appearance of the data will be affected.

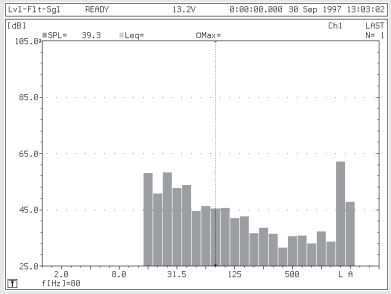
Use TYPE to switch between the display modes.

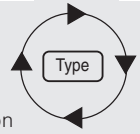
The Four Display Modes

Dual display with setup information

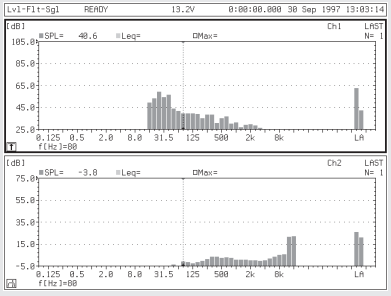


Single display (active window)

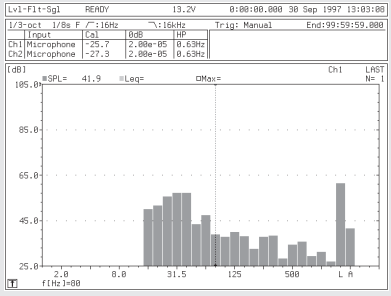




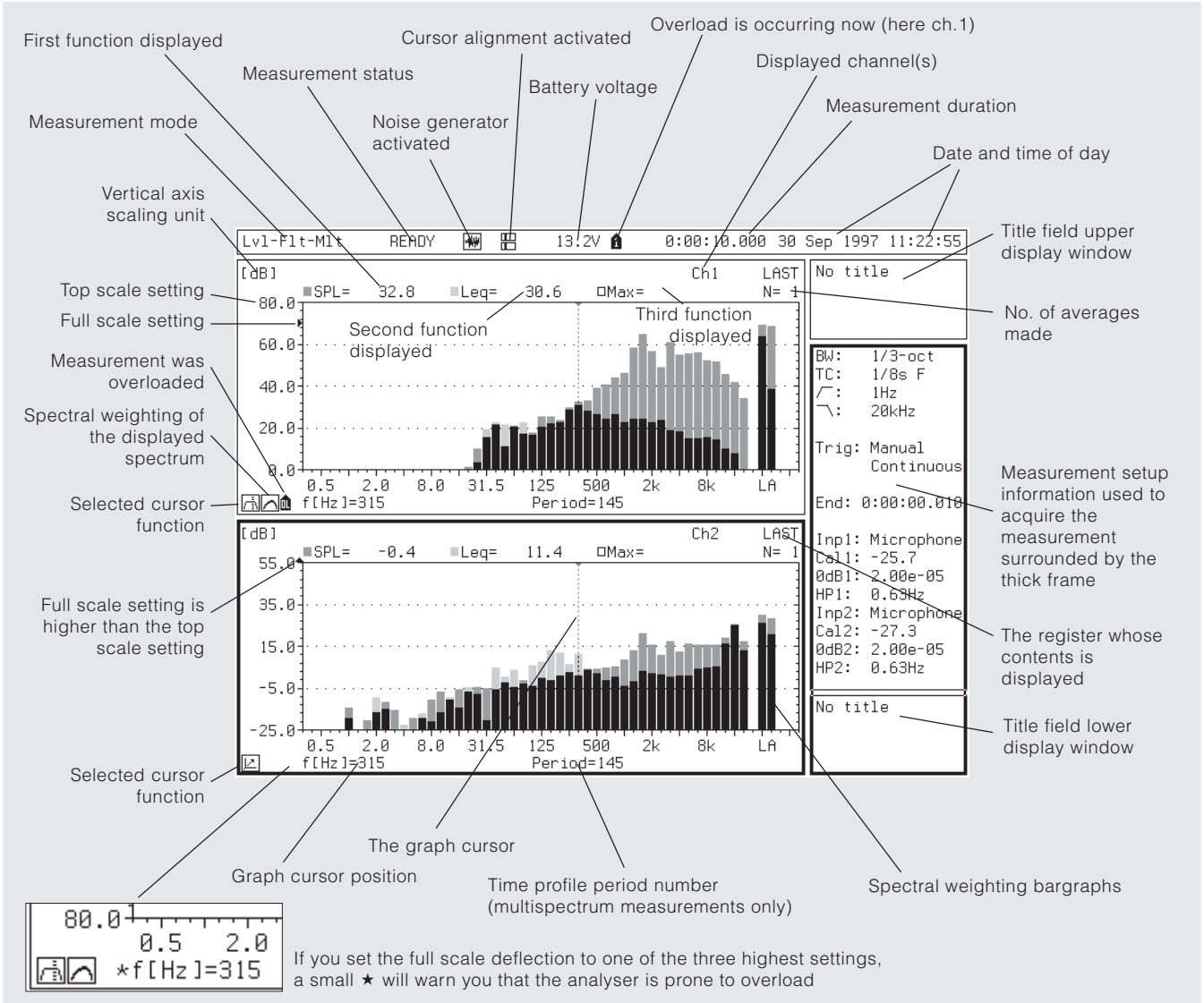
Dual display without setup information



Single display
with condensed setup information



Note: The data of the active window will be the only set of data printed out when you make a numerical printout.



If you set the full scale deflection to one of the three highest settings, a small ★ will warn you that the analyser is prone to overload

This setup menu controls the type of functions (values measured, e.g. Leq or Lmax) to be displayed, the scaling of the axes and the layout of the numerical table.

There is one setup menu for each display window and one set of setup menus for the single spectrum mode and another for the multispectrum mode – i.e. one set of two menus for each mode.

These two sets of setup menus are completely independent of each other; while you can copy the display setup of the upper window to the lower window (and *vice versa*), you cannot copy between the single spectrum and the multispectrum modes.

Press D.SETUP to produce the Display Setup menu.

Displayed Curves

The display can show up to three curves simultaneously. These three curves may be set to show any of the measured functions (if you set them to functions not measured, no curve will be displayed).

However, when you select *Ch1 and Ch2*, only the first (the left-most) setting will be applied to the display. The settings of the two others will be ignored. Two curves will be displayed – one for each channel. Which function the curves show will depend on the curve specified in parameter-field 1 of the Curves setup in the Display Setup menu. Should this parameter-field be set to *0:Off* the analyser will search for the first activated function and use that for the two curves instead.

The Numerical Table

The Display Setup menu is also used for setting up the functions to appear in the numerical table.

The table consists of six columns. If you set e.g. three columns to display the same function, this function will appear three times in the table.

In addition to the functions available for graphical display, the table also offers the option of displaying the number of averages made within each filter band (the average counter).

Note that if you select *Ch1 and Ch2* as displayed curves, the analyser will override your table setup.

The number of columns with non-zero contents will always be five, leaving the rightmost column empty. The contents of these five columns will be ch.1 data; ch. 2 data; the arithmetic difference between the contents of column 1 and 2; the average counter for ch.1 and finally the average counter from ch. 2.

The type of function displayed will be given by what you have specified in column 1 of the *Columns in the numerical table field* of the Display Setup menu.

Should Column 1 be set to 0:Off, the analyser will search for the first activated function and use that for the two curves instead.

Tip: The A- and Linear spectral weighting networks are "true" spectral weighting functions in the sense that they are separate measurement filter bands. All other spectral weighting functions are applied as postprocessing features only.

This means that if you synthesise, e.g. a B-weighting curve and fail to apply a full bandwidth to your measurement (by measuring 50–3150Hz only for example), the B-weighted values will normally deviate from those obtained with a true spectral weighting filter band. The only way to make them similar is to apply your function to measurements using a full bandwidth.

Tip: When applying spectral weighting functions other than A- and Linear (see above) the term maximum values become meaningless since we have no information on when the maximum level occurred.

Tip: Numerical (tabulated) printouts will contain the functions set active in the numerical table part of the Display setup menu only. Make sure that the functions here are functions measured, otherwise your printout will contain one or more empty columns.

The Menu for this Task (there is one independent menu for each of the two display windows) D.Setup

Defines the channel whose contents are to be displayed. Select between *ch.1, ch.2, ch.1–ch.2, ch.2–ch.1, ch.1 and ch.2, ch1–ref1, ch2–ref1, ch1–ref2, ch2–ref2, ch1+ref1, ch2+ref1, ch1+ref2 and ch2+ref2*

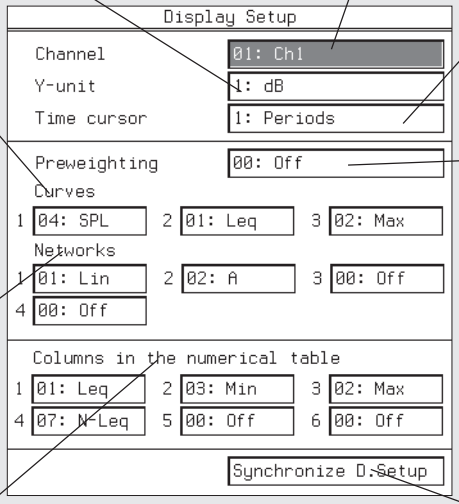
Type of unit to be used for the vertical axis. Select between *dB* and *absolute* (engineering) *units*

Time axis scaling. Applies to multispectrum measurements only. Select between *Periods*, *Relative time* (since trigger) and *Absolute time* (date and time of day)

Up to three set of curves (graphs) may be shown simultaneously. Which functions to show is defined here. Choose between *Off, Leq, Max, Min, SPL, SEL.*

In multispectrum mode, only the functions actually recorded will provide graphs

The displayed graph can be shown spectrally weighted. Options are *A, B, C, L, W1–W8**



There are up to four bargraphs located to the right of the spectrum in the display. Select which ones to appear among: *A, Lin, SumA†, SumB, SumC, SumL* or *W1–W8**

The contents of the numerical table is determined by the setup of these parameter fields. Both the numerical display and the numerical output (printout) is affected by this setup. The options to select from are the same as those applying to the displayed curves

There is one display setup menu for each of the two windows (the upper and the lower window). Instead of going to the other window and set up the display, you can apply the same setup to the other window by moving the cursor to this field and then exit the menu (by pressing ENTER)

* W1–W8 denotes the spectral weighting function you can make yourself, either by keying in the gain/attenuation values of each frequency band or by converting a measurement to a weighting curve

† The term SumA denotes the A-weighted value calculated from the measured spectrum, while the A as such is a true A-weighting filter applied to the measurement as an independent measurement channel. The two may differ for two reasons; round-off differences in the digital calculation process (normally small deviations) and if the frequency range of the measured spectrum is limited – see the tip on the left side of this page spread.

The Nor-840 has an extensive set of cursor functions. However, some apply to certain situations only.

All the cursor controls are located around the DIAL. To operate a cursor you start by selecting the type of cursor and then use the DIAL or PREV & NEXT. A small icon appears in the display to tell you which cursor function has been activated.

Scaling and Graduation

To optimise the presentation of the measured functions and values, you may adjust the horizontal and vertical axes' graduations (X-RANGE and Y-RANGE, respectively); the vertical axis top scale value (Y-MAX) and the horizontal axis minimum value (X-MIN).

Z-axis Cursor

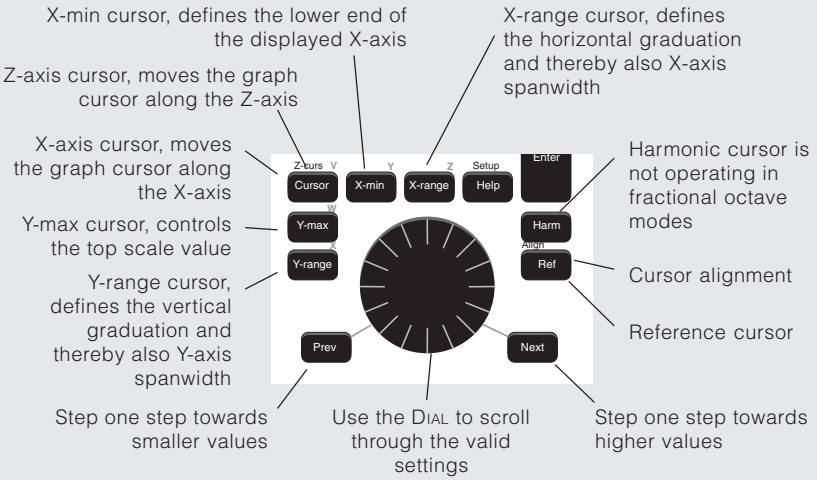
In multispectrum mode, the measured data can be represented by a three-dimensional matrix having level, frequency and time as the three dimensions.

If the display is set to display a spectrum, the X-axis will be the frequency axis, the Y-axis will be the level axis and the Z-axis will be the time axis.

You may also display the time profile for a certain frequency band (e.g. the 3150Hz 1/3 octave band or the A-weighted value). In this case the X-axis becomes the time axis and the Y-axis remains the level axis, while the Z-axis now becomes the frequency axis.

To switch between time and frequency as X-axis, use the LF/LT key.

The Cursor Control Keys


X-min cursor, defines the lower end of the displayed X-axis
X-range cursor, defines the horizontal graduation and thereby also X-axis spanwidth
Z-axis cursor, moves the graph cursor along the Z-axis
X-axis cursor, moves the graph cursor along the X-axis
Y-max cursor, controls the top scale value
Y-range cursor, defines the vertical graduation and thereby also Y-axis spanwidth
Harmonic cursor is not operating in fractional octave modes
Cursor alignment
Reference cursor
Step one step towards smaller values
Use the DIAL to scroll through the valid settings
Step one step towards higher values
Z-curs
Cursor
X-min
X-range
Setup
Help
Enter
Y-max
Y-range
Harm
Align
Ref
Prev
Next

Note that the Y-axis is left unaffected by this swopping, since the level in both cases will be the same (RMS or Peak values, always in dB).

To move along the X-axis (irrespective of whether this represents time or frequency) press CURSOR and then use the DIAL or the PREV & NEXT keys.

To move in the Z-direction (again irrespective of what it represents) press 2ND CURSOR and do as for cursor movements along the X-axis. A small icon in the display will indicate that Z-cursor is selected.

3D Cursor Functions

Measured data acquired in multispectrum mode may also be displayed as a three-dimensional graph. For obvious reasons this feature does not apply to the single spectrum mode.

To activate this feature make a multispectrum measurement, set the column 1 of the *Displayed curves* in the Display Setup menu to one of the functions actually measured and press the 2ND LF/LT keys.

If you fail to set up a function actually measured, all you"ll see will be an empty floor and no graph.

The graph is always drawn so that the periods most recently acquired appear to be closest to you.

Use the LF/LT to flip the X- and Z-axis.

Whilst the graph is being drawn the TYPE key is used to select the next display type. The CURSOR with or without 2ND, the X-MIN and the X-RANGE are used to select the cursor function, to which the DIAL and the PREV & NEXT keys apply. However, these step control keys will not cause response until the graph has been redrawn completely.

When the keys Y-MAX; Y-RANGE; 2ND X-RANGE and 2ND Y-RANGE are pressed prior to using the step control keys, any display update in progress will be aborted and a new update started.

Cursor Alignment

In dual display mode the graph cursors (X-axis cursors) of the two windows may be aligned with each other and moved together in either direction.

Press 2ND REF to align cursors. To deactivate press again, press TYPE or change to another measurement mode (e.g. single spectrum or intensity).

The master cursor will be the cursor of the active window (the one surrounded by the thick frame).

The two display windows must contain data acquired with the same filter bandwidth (e.g. 1/3 octave bands), but they need not display data in identical domains – i.e. one can show data with time as X-axis while the other shows frequency as X-axis. The contents of the two windows need not belong to the same measurement.

Reference Cursor

The reference cursor is used to investigate the difference between two points on a graph.

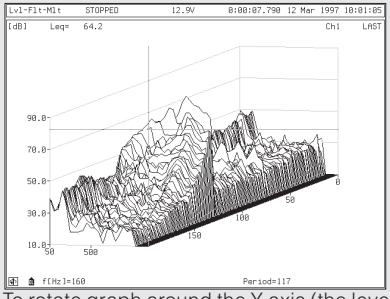
To activate the reference cursor, select CURSOR to move the graph cursor to your reference point, then press REF and use the DIAL or PREV & NEXT to move the graph cursor to a point elsewhere on the graph. The reference point will now be stated in a line inserted in the active window.

The Icons Show the Type of Cursor Selected

A B C D E F G H

A: X-min
B: X-range
C: Y-max
D: Y-range
E: Graph cursor
F: Z-cursor
G: Y-axis rotation (3D)
H: X-axis rotation (3D)

3D Cursor Functions



To rotate graph around the Y-axis (the level axis) press 2ND Y-RANGE and use PREV & NEXT or the DIAL. Origin of rotation is not origin, but a point exactly in the middle of the XZ- (time-frequency) plane.

To rotate the graph around the X-axis, press 2ND X-RANGE and use PREV & NEXT or the DIAL.

PREV & NEXT step size is 0.03 radians.

Remember to set the column 1 in the Display Setup menu to a function actually measured to get a graph.

Cursor Alignment

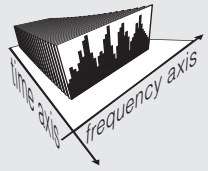
The master cursor is always the cursor of the active window (surrounded by the thick frame).

The two displays must contain data with identical filter bandwidth (e.g. 1/3 octave bands) but one can show spectrum while the other shows time profile.

The cursor will never move outside the master cursor's range, but if the period range of the master cursor window exceeds that of the other window, moving the graph cursor will cause the slave cursor to stop at its extreme end, while the master will go on until it reaches its extreme end. To spot when this occurs, watch the period number in each display window.

If the cursor alignment was active at the time you stored a setup file (a .cfg file) and you put this on a floppy disk naming the file 840.cfg, initialising the analyser with this floppy will cause the cursor alignment function to be set active as a part of the initialisation.

Tip: Do not confuse top scale and full scale! The top scale is purely a display control indicating the top of the vertical axis, while the full scale controls the input amplifier gain settings. Changing the latter will delete the Last register contents!



More 3D. If the X-axis represents the frequency, the Z-axis will represent the time and vice versa. Use LF/LT to swop.

Scaling and Graduation Range

Y-axis: 20, 40, 60, 80 or 100 dB across the vertical scale or in a 1–2–5 sequence when set to engineering units.

X-axis: 1:8, 1:4, 1:2, 1:1, 2:1, 4:1 and 8:1 applies to time displays only (not the fractional octave spectrum displays)

Data measured or retrieved can be shown either as graphs or tabulated. To switch between these two ways of presenting the data press the NUM. key.

The contents of the table is determined by the Display Setup menu.

You navigate in the table by means of the *DIAL*, PREV & NEXT and the 2ND 9 (PGUP), 2ND 3 (PGDN), 2ND 7 (HOME) and 2ND 1 (END).

The table is no more than a numeric representation of the data, so the Z-cursor and cursor alignment works even here.

A Tabulated Example

Num.

Lvl-Flt-Mlt ENDED 13.2V 0:00:10.000 1 Oct 1997 11:08:46

Period=5 Ch1 LAST

Frq[Hz]	Leq[dB]	Min[dB]	Max[dB]	N-Leq
1k	25.3	27.0	27.0	1
1.25k	27.6	26.5	26.5	1
1.6k	27.3	24.0	24.0	1
2k	20.1	23.5	23.5	1
2.5k	21.5	24.9	24.9	1
3.15k	17.7	20.4	20.4	1
4k	15.7	17.9	17.9	1
5k	16.6	17.0	17.0	1
6.3k	13.4	15.7	15.7	1
8k	14.2	17.4	17.4	1

BW: 1/3-oct
TC: 1/8s F
⌐: 1Hz
¬: 20kHz

Trig: Manual
Continuous

End: 0:00:00.010

Inp1: Microphone
Cal1: -25.7
0dB1: 2.00e-05
HP1: 0.63Hz
Inp2: Microphone
Cal2: -27.3
0dB2: 2.00e-05
HP2: 0.63Hz

f[Hz]=1k Ch2 LAST

Periods	Leq[dB]	Min[dB]	Max[dB]	N-Leq
1	0.9	-0.7	-0.7	1
2	1.9	-0.7	-0.7	1
3	24.4	0.0	13.4	1
4	5.3	13.2	13.2	1
5	1.7	13.0	13.0	1
6	-1.2	12.7	12.7	1
7	-8.0	12.1	12.4	1
8	-6.0	11.9	11.9	1
9	-2.4	11.6	11.6	1
10	1.4	11.3	11.3	1

Here the upper window has been set to display L(f) of a multispectrum measurement while the lower shows L(t) of the very same (might as well be from two different) measurements. We have activated the cursor alignment so that the two tables have the same period of the same frequency band shown highlighted.

Editing in the Tables

You may edit tabulated values except the Last register. Transfer data to another register before editing.

To start editing go to the line to edit and press 2ND AUX (EDIT). Selected position is then shown highlighted.

Use the numerical keypad to key in the new value. Terminate by ENTER or abort editing with ESC.

Use DIAL, PREV & NEXT and the 2ND 9 (PGUP), 2ND 3 (PGDN), 2ND 7 (HOME) and 2ND 1 (END) to move up and down in the table.

Use ❖ (Field cursor) – those pointing horizontally – to move between columns.

Press 2ND AUX (EDIT) to deactivate function.

When You Edit in the Tables, it looks like this...

Lvl-Flt-Sgl ENDED 13.1V 0:00:08.188 22 Sep 1997 13:05:39

Ch2 USER

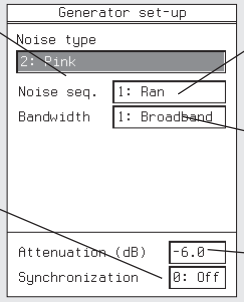
Frq[Hz]	Leq[dB]	Min[dB]	Max[dB]	N-Leq
200	13.9	4.8	36.5	1
250	16.6	7.4	37.7	1
315	18.2	7.3	37.1	1
400	20.0	8.0	37.9	1
500	27.0	5.7	45.1	1
630	27.7	7.4	46.7	1
800	22.0	10.8	40.6	1
1k	23.4	11.7	41.5	1
1.25k	23.7	12.8	41.3	1
1.6k	21.0	13.7	37.8	1
2k	19.4	15.0	35.3	1
2.5k	23.6	15.5	40.6	1
3.15k	23.6	17.7	40.3	1
4k	34.9	34.4	41.7	1

The Menu for this Task

2nd Gen

The noise type is
1:White
2:Pink
3:BP-filtered
4:Impulse
5: BP-filtered impulse

Lets the generator span (or follow) the measurement. When set to 1:On and START is pressed the noise generator will be turned on before the measurement is started and switched off after the measurement has ended

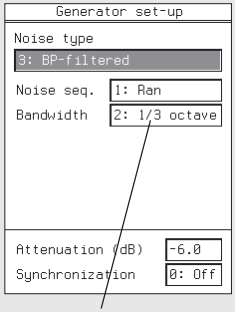


The noise sequence can be either *1:Ran*(dom) or *2:P*(seudo)*Ra*(ndom)

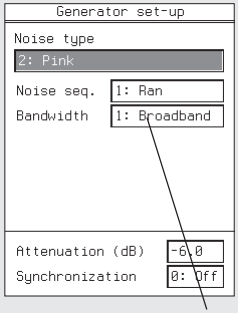
Extra field to define bandwidth when applicable – see below

Set the noise generator output level in dB re: 1V

Range: –40.0 dB to 0.0 dB in 0.1 dB steps



If you select bandpass filtered noise, you must specify the filter bandwidth (1/1 or 1/3 octave bands)



If you select pink noise, you must specify whether broadband (20–20 000 Hz) or limited (100–5 000 Hz) shall be used

This icon appears in the status line of the display when the noise generator is running

The Nor-840 comes with a powerful noise generator built in.

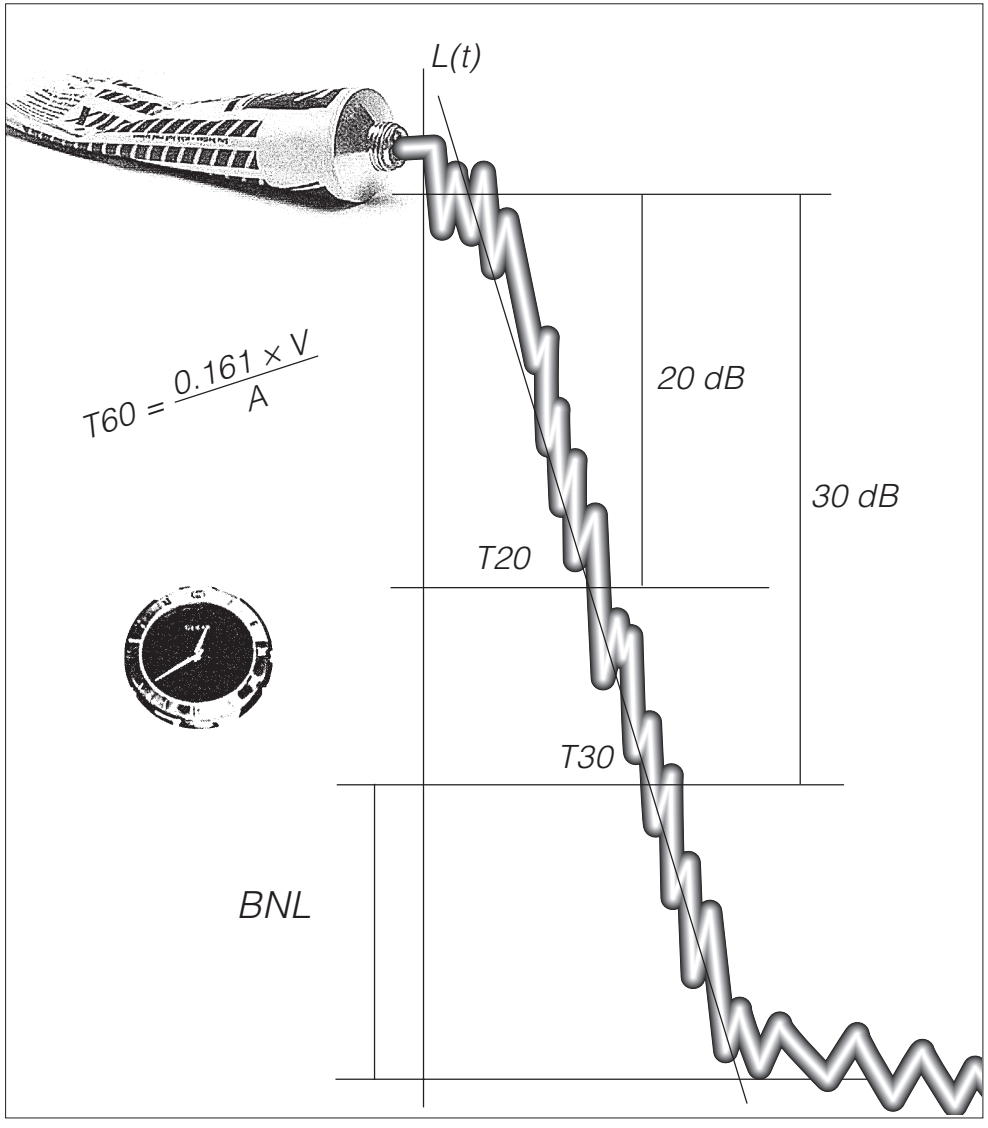
To access the setup menu press 2ND GEN.

The generator can be set up to supply white noise – whose level will increase by 3 dB per octave when viewed using fractional octave filters; pink noise which looks flat in the fractional octave domain; bandpass filtered (1/1 or 1/3 octave bands); impulse and bandpass filtered impulse.

For the bandpass filtered signals, the frequency (band) is determined by the current graph cursor position.

Once the generator has been properly set up and the setup menu closed, press GEN to activate the generator and again to deactivate.

36 Reverberation Time Measurement Fundamentals
38 Setting up for Decay Captures
39 Calculating the Reverberation Time
40 Averaging and Viewing the Calculated Values



Making Reverberation Time Measurements

Reverberation time (RT) is defined as the time it takes for the sound pressure level in a room to decay by 60 dB when no more sound is fed into the room.

However, the presence of background noise will normally prevent you from measuring a full 60 dB decay. The normal circumvention is then to measure decays of 20 or 30 dB and extrapolate the results to 60 dB.

In the Nor-840, we specify the actual decay range over which the measurement took place by the annotation T_{20} and T_{30}.

An ideal sound decay will form a straight line when drawn in a coordinate system with logarithmic axes. In reality, however, sound decays will always contain fluctuations. Two problems will then immediately arise; viz. how to have the analyser accurately determine the initial level and when to start the calculation.

Noise Excitation

With noise excitation, the calculation starts at 5 dB below the mean level (i.e. the L_{eq}) of the noise measured at the microphone position before the noise is switched off.

As long as the noise stays below this –5 dB line (the L_1 in the Fig.) the time elapsed is counted. Once the level drops below a second line (the L_2 in the Fig.) the counting is discontinued. Should the level, for any reason again exceed the second line, the counting will be resumed and go on until the level once again drops below this line.

Likewise, should the level exceed the first line any time after counting has started, the counting will be discontinued until the level once again drops below this line.

The second line will be 25 dB below the initial level for T_{20} measurements and 35 dB below for T_{30} measurements, leaving a measurement range of 20 and 30 dB respectively.

To reduce the influence of fluctuations at the start and the end of the time measurement, a triangular weighting function is applied to the decay. More emphasis is then applied to the middle part of the decay and less to its extremes.

Impulse Excitation

For impulse excitation, the calculation starts at 5 dB below the maximum level measured.

As long as the sound level stays below the –5 dB line, the time elapsed is counted until the level is 25 dB below maximum level measured for T_{20} measurements and 10 dB further down (35 dB altogether) for T_{30} measurements.

The main difference between the noise and the impulse excitation calculation method lies thus in the way the initial –5 dB level is determined.

Level Mode Is Used for Acquisition

Although the RT Mode is a separate measurement mode it uses the multispectrum Level Mode to acquire the decays. Reverberation times can be calculated on the basis of level mode measurements only – FFT and Sound Intensity won't work.

As a separate measurement mode, the RT Mode has its own set of registers, of course, independent of the Level Mode registers.

The RT calculation procedure is based on excitation type information, Level Mode multispectrum period length information and the preset requirement for minimum distance to the noise floor (see below).

Hence, the RT Mode becomes unusable without the multispectrum Level Mode, yet it has influence on the setup of the multispectrum measurement via the period length setting and when *Revert to default* is activated, it also controls the trigger condition setting and the range of frequency bands activated.

Minimising Background Noise

You may specify a "signal-to–noise" ratio requirement for your measurement to minimise any possible influence from the background noise level.

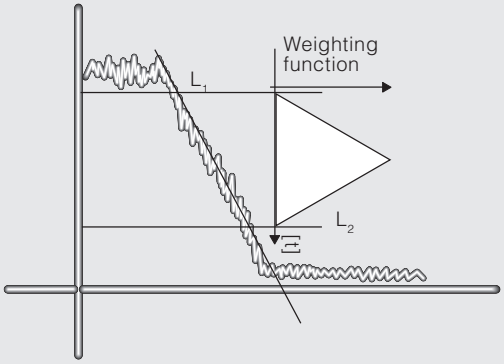
This is called the *minimum distance to the noise floor* and serves to avoid that a sudden increase in the background noise level causes the reverberation time to seem longer than it actually is.

If the minimum distance to noise floor requirement has been violated, a warning will be produced. In the numerical table, a "?" will appear next to every frequency band affected.

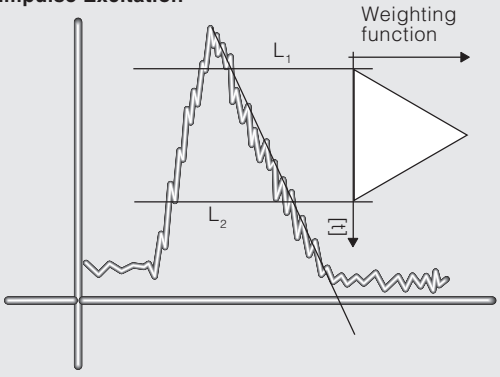
The minimum distance to the noise floor can be set to anything in the range 0–30dB, both extremes included. Resolution is 0.1 dB.

Although some International Standards such as the ISO 60 354 requires a minimum distance of 15 dB (a fairly tough requirement actually), a minimum distance of 5 to 10 dB will suffice in most cases. Default setting is 5 dB.

Noise Excitation

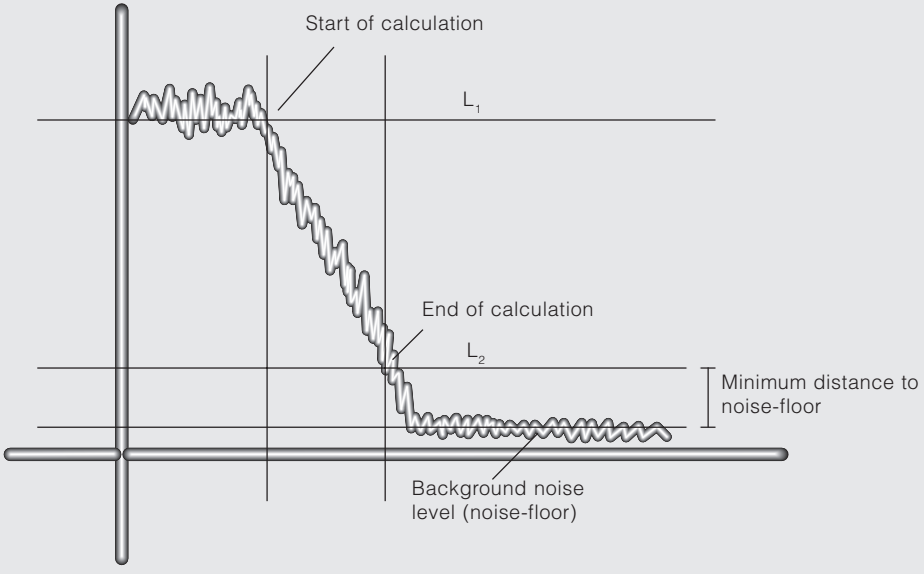


Impulse Excitation



A triangular weighting function is applied to the the counting of time elapsed to minimise the effect of fluctuations in the extremes.

Some Vital Terminology



With noise excitation, the calculation starts at 5 dB below the mean level (i.e. the Leq) of the noise measured at the microphone position before the noise is switched off.

With impulse excitation, the calculation starts at 5 dB below the max. level detected.

As long as the noise stays below this –5 dB line (the L_1 in the Fig.) the time elapsed is counted. Once the level drops below a second line (the L_2 in the Fig.) the counting is discontinued. Should the level, for any reason again exceed the second line, the counting will be resumed and go on until the level once again drops below this line.

Likewise, should the level exceed the first line any time after counting has started, the counting will be discontinued until the level once again drops below this line.

Setting up for Decay Captures

You may either use a single channel for reverberation time (RT) measurements or you may use both channels and later combine the results into one channel. The latter lets you cover twice as many microphone positions and thereby reduce significantly the time spent on creating spatial averages of the RT.

If you prefer to use one channel only for your RT measurements, you should turn off the channel you won't be using. You can do this in the *Input* menu.

Revert to Default will, irrespective of excitation type, set…

- Bandwidth to ⅓ octave bands
- No. of periods to 1000 (200 before trigger)
- Trigger to repetitive, No delay
- Measurement to end at measurement time
- Lower freq. to 50 Hz, Upper to 10 kHz
- Values to record to Leq only
- Follow cursor to Disabled
- Minimum dist. to noise floor to 5 dB
- Time constant to F
- Period length to the same value as Max RT expected, but in msec.

In addition, for Impulse excitation…

- Trigger to Level above 30 dB below full scale setting at1 kHz in ch.1 (ch. 2 if ch.1 is set to Off)
- Backwards integration mode to On

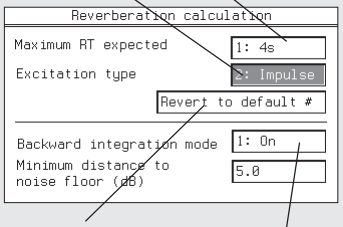
In addition, for noise excitation…

- Synchronisation to Off
- Trigger by Noise Off
- Noise generator to Pink and Ran(dom)
- Excitation time to half the setting of the Maximum RT expected

The Menus for this Task

Maximum reverberation time expected. Options: *4; 8;16* and *32* seconds

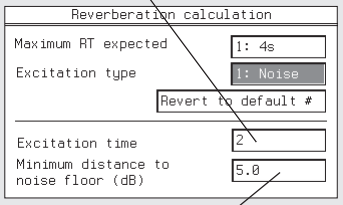
Excitation type *(noise or impulse)*



Revert to the default setup

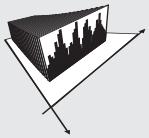
Backward integration mode *On/Off* (Impulse excitation only)

When using noise excitation the noise generator on-time (for the internal noise generator) must be specified (in seconds)

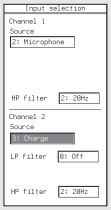


The minimum distance to the noise floor serves to minimise the influence of the background noise

Setting up for the Capture

❶ 

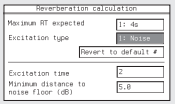
Enter Level multispectrum mode

❷ 

Press INPUT to select input source, if applicable

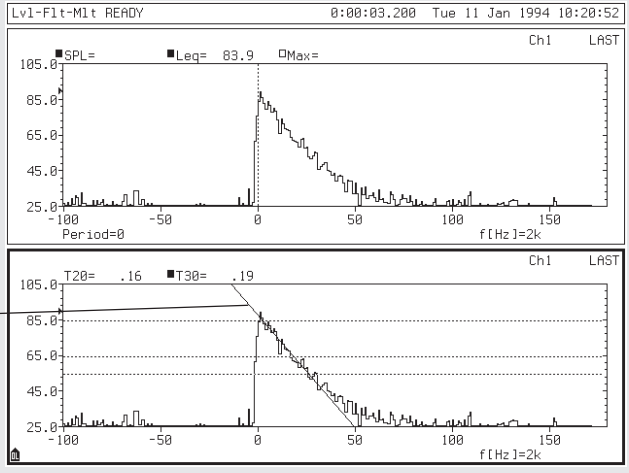
❸

Adjust full scale deflection, if needed (GAIN1/ GAIN2)

❹ 

Press 2ND RT, select excitation method and adjust the other settings as required

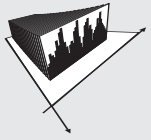
Viewing the Decay



The straight line appears for your information only, the slope cannot be changed by the user! However, you may edit the values!

Capturing the Decays and Calculating the Reverberation Time

❶



Enter Level multispectrum mode and set up inputs, full scale etc.

❷

Start the measurement.

Switch on the noise generator (noise excitation selected) If you activated revert to default the generator will start by itself.

If impulse excitation was selected, generate the impulse.

❸

Transfer the acquired data to the AVRG register (clear it before using it the first time), move the microphones and make another measurement.

Repeat for as long as needed.

❹

Switch to view the AVRG register and press RT key to begin calculating the reverberation time.

You may save your captured sound decays on the hard disk or a diskette. Observe that you do not need to calculate the reverberation time before you save the decays – the calculation can in principle be applied to any multispectrum measurement, those just acquired as well as those stored. However, calculations made on non-decaying signals will not yield meaningful results.

There are restrictions on the lower limit of reverberation times – if the decay is to steep, you will instead measure the impulse response of the filters!

In the Appendices section of this manual, there is a set of tables listing the minimum reverberation times allowed.

If your analyser is equipped with the optional Maximum Length Sequence (MLS) extension, you may utilise this mode's time reversal option which opens up for calculations of even shorter reverberation time values.

Averaging and Viewing the Calculated Values

Most applications of reverberation time (RT) calculations call for averaging to rule out local phenomena not representative for the measurement site as such.

Basically, you have two ways of creating averaged RT values; viz. by averaging several decays and *then* calculate the RT values or by averaging several individual RT values.

Although either method may be used, leading institutions such as the PTB in Germany recommend the averaging of decays rather than the averaging of individual RT values.

Applying the Schroder Method

If you use impulse excitation, the method of decay-averaging lets you apply the *backward integration method* (aka *the Schroder method*) as an afterthought, without destroying the original data.

Press 2ND followed by RT and set the *Backward integration mode* to *1: On*.

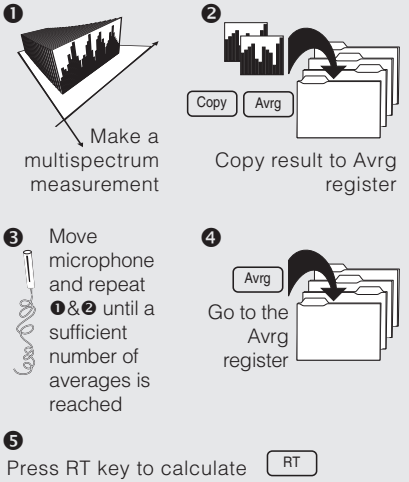
You will be warned that the RT data register will be cleared. Hence you must press the RT key to again calculate the RT values. The values calculated now will be with the *backward integration method* employed.

This afterthought works both ways, of course. If you did use the *backward integration method* you may also recalculate the RT without this method by changing the *1: On* to *0: Off*.

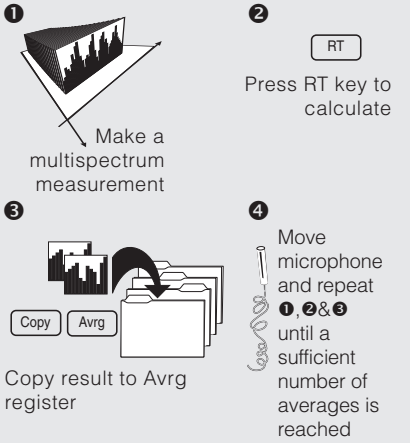
Viewing the Calculated Values

Once the reverberation time has been calculated, you have several options available to get an optimum view of the reverberation time values – see table for details.

Averaging Decays Before Calculating RT

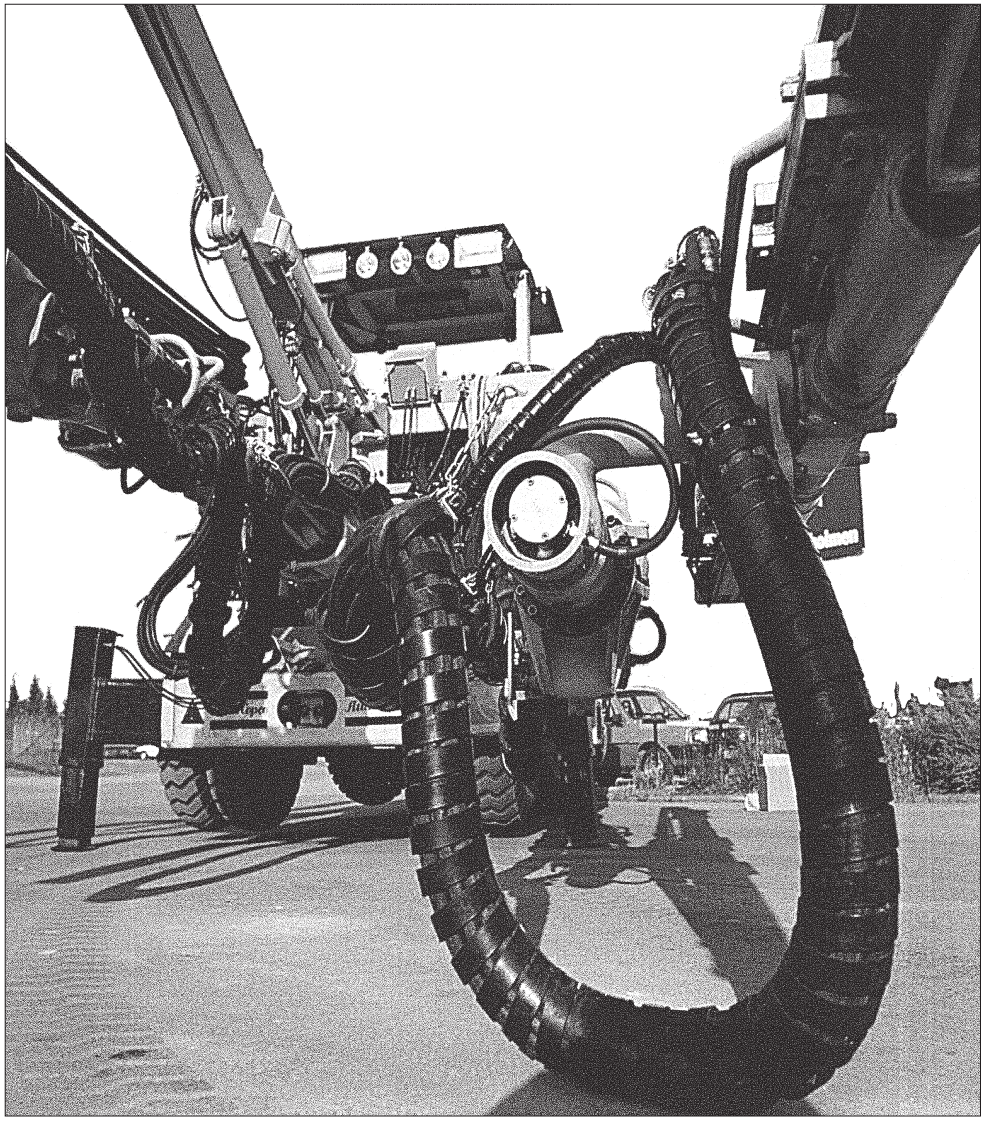


Calculating RT Before Averaging



To achieve this:	Do this:
View the decay and RT profile of another frequency band	Select 2nd Cursor as cursor function
Change the horizontal graduation	Select X-range as cursor function
Change the horizontal datum (lower end of scale)	Select X-min as cursor function
Change the top scale value	Select Y-max as cursor function
Change the vertical graduation	Select Y-range as cursor function
Switch between display modes	Press the Type key a number of times until the required display mode is reached
Switch between display of RT-related and multispectrum-related graphs	Press the RT key. No new calculation will be made until changes are made in the Reverberation calculation menu
Switch between display of a decay and RT vs. frequency	Press the Lf/lt key
Display a tabulation of the calculated reverberation time values	Press the Num. key

Chapter 4



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Making FFT Measurements

Fast Fourier Transform (FFT) is a means of calculating the spectrum of a time function. It is a digital process, using a signal's amplitude at discrete moments in time to produce the corresponding spectrum which also will consist of discrete frequency lines (a non-continuous spectrum).

A time buffer consisting of 2048 samples representing the time history of a signal is acquired. The length – in seconds – of the buffer and thereby also the spacing between the time samples depends on the upper frequency of the analysis. The higher the upper frequency, the shorter the time buffer length will be, measured in seconds. It will, however, always consist of 2048 samples.

The FFT process generates 2048 frequency lines based on the 2048 line time buffer. Out of these 2048 lines, only 1024 are unique and distributed evenly over the frequency range used.

Furthermore, out of these 1024 unique lines we use 801 lines since those affected by the anti-aliasing filter of the analyser must be discarded. To be precise we use 800 lines + 1 for DC. In normal operating mode the DC line is not in use since the analyser is AC-coupled. It is, however, used in zoom mode.

When we talk about time signals in the following, we are always referring to the contents of time buffers.

Auto Spectrum

It can be shown that the spectrum calculated will be complex – i.e. it will contain real and imaginary values. This is just a function of the moment of observation.

Let X(f) be the complex spectrum of the signal x(t) and X*(f) its complex conjugate.

The product

$$G_{11}(f) = X(f) \times X^*(f)$$

will then be real, maintaining the amplitude (the length of the complex vector) of the X(f). We have thus obtained a signal whose amplitude is independent of the moment of observation.

This spectrum, the $G_{11}(f)$ is called the *auto spectrum* of the x(t).

Cross Spectrum

Let x(t) and y(t) denote the two input signals to a dual channel FFT analyser. Let X(f) and Y(f) denote the corresponding spectra.

The product

$$G_{12}(f) = X(f) \times Y^*(f)$$

is then called the *cross spectrum* of x(t) and y(t). By taking the complex conjugate of one of the two, the only phase information present will be the phase difference between the two signals. Again we have escaped the dependence of the moment of observation.

Frequency Response Functions

Assume that we have a system with one input and one output. Let x(t) denote the input time signal, y(t) the output time signal and X(f) and Y(f) the corresponding spectra.

The following relationship exists:

$$Y(f) = H(f) \times X(f)$$

in which H(f) stands for the frequency response function (the transfer function for sinusoidal signals).

If we multiply both sides of the equation by X*(f) the equation now yields:

$$X^*(f) \times Y(f) = H(f) \times X^*(f) \times X(f)$$

which should be recognised as

$$G_{XY}(f) = H(f) \times G_{XX}(f)$$

giving

$$H(f) = \frac{G_{XY}(f)}{G_{XX}(f)} = H_1(f)$$

Similarly it can be shown that

$$H(f) = \frac{G_{YY}(f)}{G_{YX}(f)} = H_2(f)$$

and that

$$G_{YY}(f) = |H(f)|^2 \times G_{XX}(f)$$

Are there cases where the three ways of estimating the frequency response yield different results? The answer to this is clearly yes. The latter of the three clearly stands out being based as it is on the two auto spectra and hence it contains no phase information.

$H_1(f)$ tends to be preferable when the extraneous noise in the system occurs predominantly at the output of the system, while $H_2(f)$ on the other hand, is preferable when the noise is predominantly at the input of the system. Both the $H_1(f)$ and the $H_2(f)$ are available as selectable functions in the FFT extension of the Nor-840.

Time Weighting – Windows

The FFT process assumes that the time record buffer contains one period of a periodic signal. Although this is in general not true for the signals we're measuring on, it is a requirement needed to make the FFT work.

Now, if the two extremes of the time buffer have very different amplitude, connecting the end of one buffer to the beginning of the next (because of this artificial periodicity which is needed) will introduce a discontinuity not present in the original signal – see Fig. to the right for details.

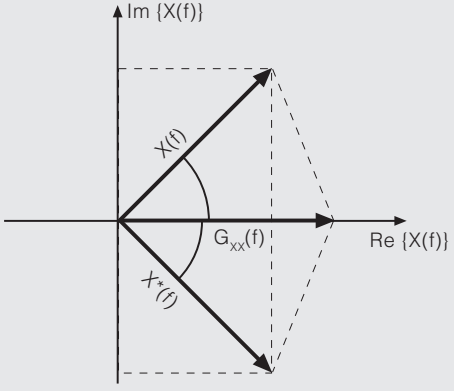
This discontinuity is very steep and will appear as false frequency components and/or incorrect levels in parts of the spectrum. False and incorrect in relation to the signal itself, of course, there is nothing false about the process as such.

There is one way to avoid this, and that is to ensure that the two extremes of the time buffer always have identical values. The easiest value is 0 (zero). Therefore we force the value to become zero in both extremes of the time buffer. By doing it gradually and smoothly, we ensure that no abrupt transitions are introduced.

This means that we put more emphasis or weight on some parts of the time buffer (the middle part) and less on others (the extremes). The technique is therefore often referred to as time weighting or windowing. If the signal contained in the time buffer is shorter than the buffer length, no weighting will be needed.

To enter FFT mode, press the FFT key. There is no multispectrum mode for FFT measurements

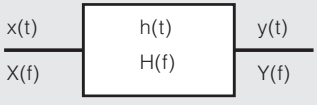
The auto spectrum is calculated from the spectrum and its complex conjugate…



By multiplying the X(f) by its complex conjugate we remove the influence from the moment of observation (i.e. the phase of the signal).

For cross spectrum the product X(f) × Y*(f) will produce a vector whose phase is identical to the difference between the phases of the two vectors and again we have removed the influnce of the moment of observation.

A system with one input and one output…



We use the two auto spectra and the cross spectrum to obtain the frequency response of the system. In this way the phase relationship is maintained – see text for details.

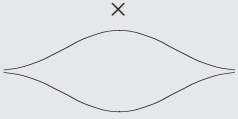
The time buffer is considered as one period of a periodic signal…

If the two extremes are not identical in amplitude a discontinuity not present in the input signal is created when the time buffers are (artificially) put after each other.

To avoid the discontinuities at the extremes of the time buffer (Fig. lower left) the amplitude at the extremes are forced to identical values – zero…

The time buffer contents…

is multiplied by a time window, here shown bipolar to clarify…



…and the result is zero at the extremes!

There are many different windows functions around, but Hanning (with the shape of a cosine) is the most commonly encountered.

As described in the article on the previous page spread, the FFT process assumes that the contents of the time buffer is one period of a periodic signal.

In order to avoid introducing signal components not present in the original signal the two extremes must be set to the same value (thus avoiding discontinuities at the extremes). For practical reasons this value is always set to 0.

The process of time weighting the time buffer contents is also called windowing, since we change the "window" through which we observe the signal.

Obviously, the fact that we do apply a weighting to the signal will influence its shape. However, the alterations introduced are well controlled and can to some extent be compensated for. For example, the amplitude of the windows is calculated so that the energy contents of the signal remains the same after the window has been applied.

Guidelines for the Use of Windows

In general the following guidelines are recommended for proper use of windows:

For the analysis of continuous signals:

Rectangular weighting should only be applied when analysing special sinusoidal signal – i.e. multi-sine signals having frequencies coinciding with the lines in the FFT spectrum and no frequency components elsewhere.

Hanning weighting is a general purpose weighting function and is in general the recommended window for continuous signals. Since the Nor-840 does all the analysis using 67% overlap when Hanning is applied no loss of data will occur.

Rectangular and *User-defined* cannot be recommended for continuous signals.

For the analysis of transient signals:

Rectangular weighting is the general purpose window for this application, provided that the transient is shorter than the time buffer length.

User-defined weighting may be used for short impulses and transients, mainly to improve the signal-to–noise ratio and for gating purposes.

Exponential weighting should be applied to transients that do not decay sufficiently within the time buffer length.

Hanning weighting can be applied to transients much longer than the time buffer length.

For frequency response measurements:

User-defined weighting when the input signal is impulsive.

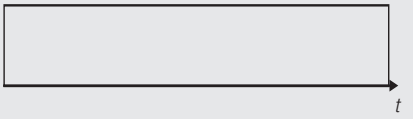
Exponential weighting for the response of the system when excited with an impulse, provided that the system is lightly damped. For zoom analysis Hanning weighting should be used instead.

Hanning weighting should be used (in both channels) when a random excitation of the system is made.

Rectangular weighting should be applied only when multi-sine signals having frequencies coinciding with the lines in the FFT spectrum are used.

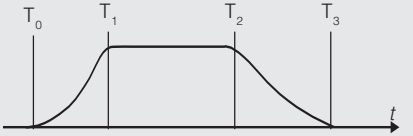
Thw Window Functions Available

a) Rectangular (no weighting at all)



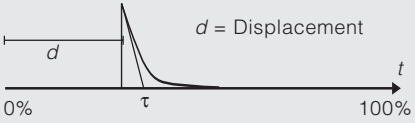
b) Hanning weighting which is a shifted cosine function starting and ending on 0

c) User-defined, which is a cosine tapered rectangular window



The four T-values indicate the start and stop of the tapering in % of the time buffer length. If T_0= 0, T_1= T_2= 50 and T_3= 100, the window becomes a Hanning window and if T_0= T_1= 0 while T_2= T_3= 100, it becomes a rectangular window

d) Exponential window



The position and size of the exponential window is given by its displacement relative to the origin (0%) and its time constant (τ)

The FFT process normally produces a spectrum covering a range from 0 to some chosen maximum frequency – the f_{MAX}. The resolution of the spectrum is determined by the size of the transform, i.e. the number of samples used to describe the time signal.

The Nor-840 has a 2048 samples time buffer which gives 1024 frequency lines evenly distributed over the range from 0 to the sampling frequency. Out of these 1024 lines we end up using 801 lines only, since those affected by the anti-aliasing filter must be discarded. To be precise we use 800 lines + 1 for DC. In normal operation the DC is not used since the inputs of the Nor-840 are AC-coupled, but when zooming you will get 400 lines on each side of the zoom centre frequency.

The line spacing is then given by

$$\Delta f = f_{MAX}/800, \text{ or } \Delta f = f_{sampling}/1024$$

This leaves us with only one option if we want to increase the resolution (corresponding to a reduction of the Δf), and that is to increase the size of the transform.

However, if we make a sacrifice, we can still increase the resolution without increasing the transform. We may increase the frequency resolution, but only within a correspondingly smaller part of the original frequency range at a time. This is called zoom FFT.

What we actually do is that we shift (by heterodyning) and then low pass filter the part of the frequency range that we are going to look at more closely. This means that the desired part of the spectrum – hereinafter referred to as ΔF – is shifted down in frequency to make it appear as if it were a signal spanning from 0 Hz to 0 + ΔF Hz.

The signal can now be analysed as if it were a signal with a low upper frequency, having a higher resolution (Δf).

Example: Assume an initial frequency range of 25 kHz. This gives a Δf equal to 31.25 Hz since 25 000/800 = 31.25. If we now decide to make a 16 times zoom over a selected range, the bandwidth will be 25 000/16 = 1562.5 Hz and the resolution will be 1.95 Hz.

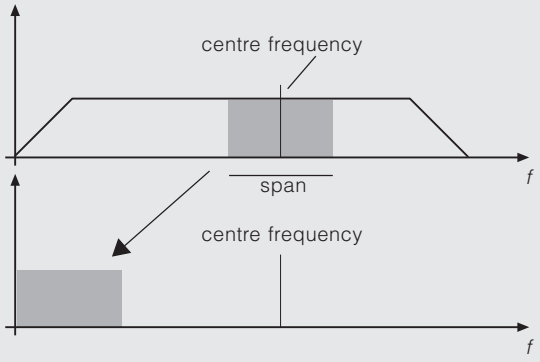
Since the bandwidth will be analysed using all the 2048 time samples, the time buffer length (when measured in terms of seconds) will increase to 16 times the non-zoomed length. Why? Simply because time and frequency is related through $\Delta f \times T = 1$, in which Δf is the frequency resolution and T is the time buffer length.

For the above example the time buffer length will be 32 ms in the initial (25 kHz) situation and 512 ms in the zoom situation.

How the Zoom FFT Terms Relate to Each Other…

Define the zoom area by entering the centre frequency and then select the zoom factor. As the Fig. shows the span must be chosen so that half its width is less or equal to the centre frequency chosen. Otherwise the span would contain negative frequencies.

The zoom process shifts the zooms range by means of heterodyning and low pass filtering. Thus the zoom analysis becomes similar to an analysis with a lower upper frequency limitation.



The time buffer will not become meaningful for the zoom FFT analysis case

Zoom FFT in the Nor-840

The zoom FFT should be used as follows:

1. Start by specifying a centre frequency (e.g. 10 416 Hz and a zoom frequency span (1 of 8 preset zoom factors)
2. Since the FFT analysis is an 801 line process, you'll get 400 lines on each side of the specified centre frequency.
3. The centre frequency must be chosen so that the entire span is contained within the frequency range. Example: A centre frequency of 1 000 Hz and a span of 3 125 Hz is not possible. However, the analyser keeps track of this and adjusts the zoom factor (i.e. the span) whenever a conflicts arises.
4. Based on your selections the analyser will display the corresponding upper and lower frequency of your zoomed frequency range.

The zoom feature is located in the measurement setup menu.

Selecting Input Source for FFT Measurements

The Nor-840 has four different types of signal input sockets. These are accessed via the input menu. Press the INPUT key to produce this menu.

Only one input can be selected at the time per channel, but the two channels need not be set to the same type of input source.

One of the channels may be set to *Off*, but both channels cannot be set to *Off* at the same time.

If you set one channel to *Off* in the input source menu, the corresponding part of the calibration menu will be blank.

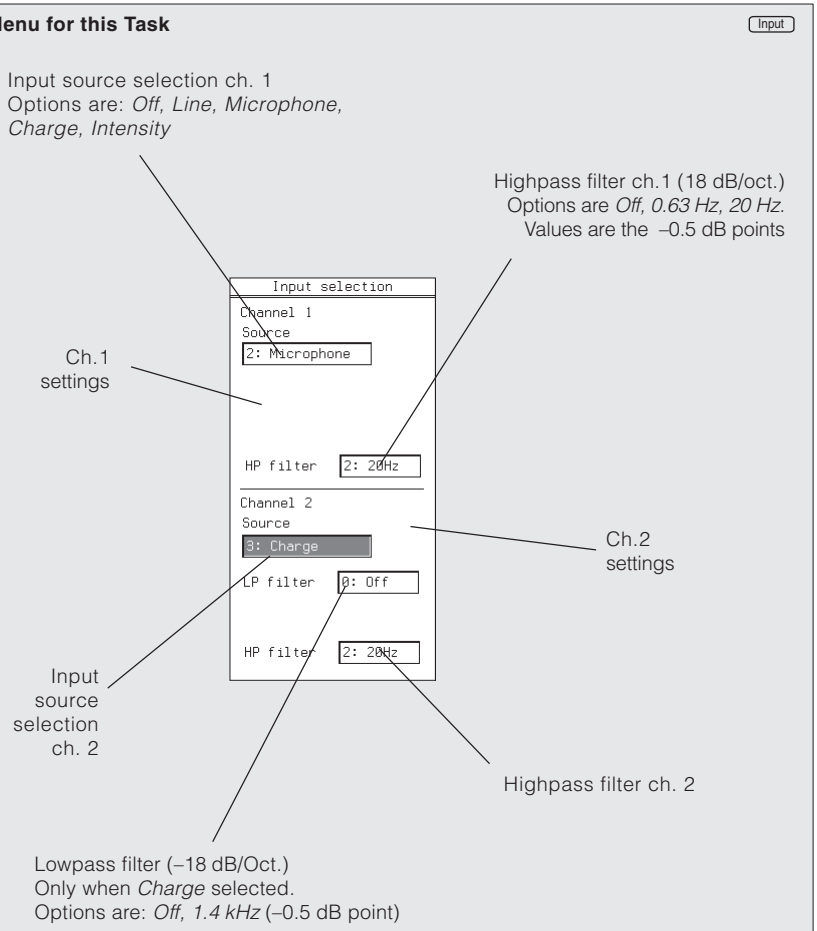
Observe that if you select sound intensity for one of the channels, the other channel will be set to the same mode automatically, since sound intensity is a two-channel measurement technique. If you then change the setting of one of the channels, the other will go back to the setting it had at the time intensity was selected.

A lowpass filter will be activated when *3: Charge* is selected.

The menu contains a highpass filter for each channel.

The highpass filter cutoff frequency stated is the –0.5 dB frequency point (not the usual –3 dB frequency point).

The Menu for this Task

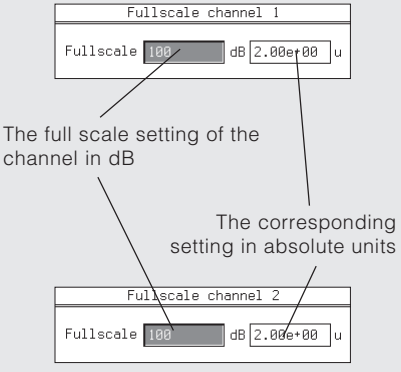


Note: If a channel is set to *Off* in this menu, the corresponding part of the Calibration menu will appear blank.

The Menu for this Task

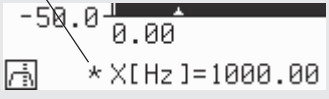
Gain1
Gain2

There is one menu per channel


Fullscale channel 1
Fullscale 100 dB 2.00e+00 u
The full scale setting of the channel in dB
The corresponding setting in absolute units
Fullscale channel 2
Fullscale 100 dB 2.00e+00 u

Do not confuse the full scale setting and the top scale setting! The former defines the input amplifier gain and hence the overload margin for a given signal level, while the latter is used to set the display to make the measured signal fit within the setting of the axis.

If you set the full scale deflection to one of the three highest settings, a small * will warn you that the analyser is prone to overload...



Note: Autoranging is not available in FFT mode

The available full scale deflection of a measurement is given as a combination of the input amplifier gain setting and the calibration setting.

The gain is set separately for each channel and is adjustable in steps of 5 dB.

Press GAIN1 (GAIN2) to enter the menu.

The full scale deflection controls the setting of the input amplifiers. However, it has no influence on the vertical scale in the display. The two extremes (top and bottom) of the vertical scale is controlled by the *top scale value*, which has its own dedicated key called the Y-MAX key. The top scale setting is purely a display function having nothing to do with the input amplifier whatsoever.

To keep you informed about the present setting of the full scale deflection a small arrow is used as full scale deflection indicator – see *A Tour of the FFT Mode Display* for details.

Note that if you set the full scale deflection to a very high value, you face the risk of having severe overload in the transducers without seeing any trace of it in the analyser – simply because the analyser input isn't overloaded. For example; a ½" microphone with a sensitivity of 50mV/Pa will distort severely (more than 3% total harmonic distortion) when exposed to levels above 135 dB(A). To warn you about this the three highest settings provide an asterisk (*) in the display.

Calibrating for FFT Measurements Is Done in Level Mode

The calibration for FFT level measurements must be carried out in Level mode.

Press LEVEL to switch to Level mode.

Although you may calibrate by just keying in the sensitivity, we always recommend that you use a calibrator. This is the only way to ensure proper operation of the entire measuring chain including the transducer(s).

If you are going to use the instrument for vibration measurement it may be convenient to change the 0 dB level to obtain dB readings easy to compare with other vibration measurements. Check with relevant Standards and conventions to find suitable or commonly used 0 dB levels.

Note that if you set the 0 dB level to a value different from 2×10^{-5}, the selected setting will appear in reverse video on the screen as shown in the lower Fig. to the right.

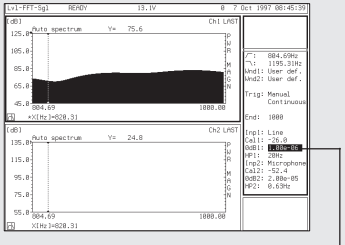
Half of this menu may occasionally appear blank. This will take place whenever the corresponding channel input has been set to *Off*.

Press the FFT key to return to FFT mode.

The Tool for this Task

Norsonic offers three different sound calibrators (available separately) to cover the requirements of the IEC 942 class 0L, class 1 and class 2. The model shown here is our class 1 calibrator, the Nor-1251. Note the adaptor designed to permit the use of 1″ as well as ½″ microphone cartridges.

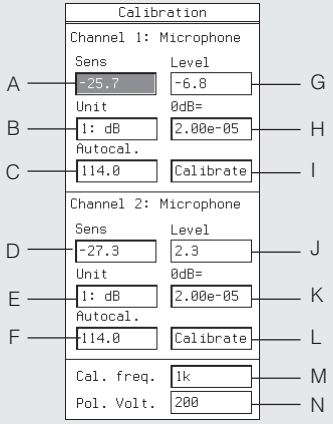
If You Change the 0 dB Reference Level



...an indication is provided

The Menu for this Task

Cal



A Sensitivity of channel 1
B Ch. 1 units (dB or engineering units)
C Auto-calibration level of channel 1
D Sensitivity of channel 2
E Ch. 2 units (dB or engineering units)
F Auto-calibration level of channel 2
G Level of ch. 1 measured with the selected time constant
H 0 dB level of ch. 1
I Initiate auto-calibration of ch. 1
J Level of ch. 2 measured with the selected time constant
K 0 dB level of ch. 2
L Initiate auto-calibration of ch. 2
M Calibrator frequency
N Polarisation voltage setting

Calibration, Using a Sound Calibrator

❶

Insert microphone into calibrator

❷

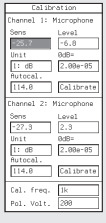
Press INPUT and select input source

❸

Adjust the full scale setting (GAIN1/GAIN2)

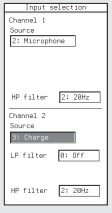
❹

Press CAL and set the calibrator frequency

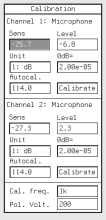
❺ 

Set the sensitivity of the microphone used until the sound pressure level is indicated correctly

Calibration, Setting the Sensitivity

❶ 

Press INPUT and select input source

❷ 

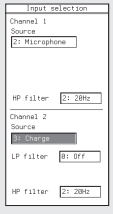
Press CAL and set the sensitivity of the microphone used

This method is in general not recommendable since it does not take into account whether the microphone actually works as it is supposed to do. Bearing in mind that the microphone itself is the most vulnerable part of the measurement chain, this method obviously suffers from severe shortcomings.

Auto-calibration Using a Sound Calibrator

❶

Insert microphone into calibrator

❷ 

Press INPUT and select input source

❸

Adjust the full scale setting (GAIN1/GAIN2)

❹

Press CAL and set the calibrator frequency

❺

Set the calibrator's output level

❻

Select *Calibrate* and press ENTER to commence auto calibration

Trigger Conditions in FFT Mode

There are separate trigger setups for each of the measurement modes of the Nor-840. Hence, the FFT mode has a trigger setup menu of its own.

The menu contains context sensitive elements; if you select trigger conditions related to amplitude, extra parameter fields will appear to let you define what amplitude and which polarity in which channel to serve as the trigger condition.

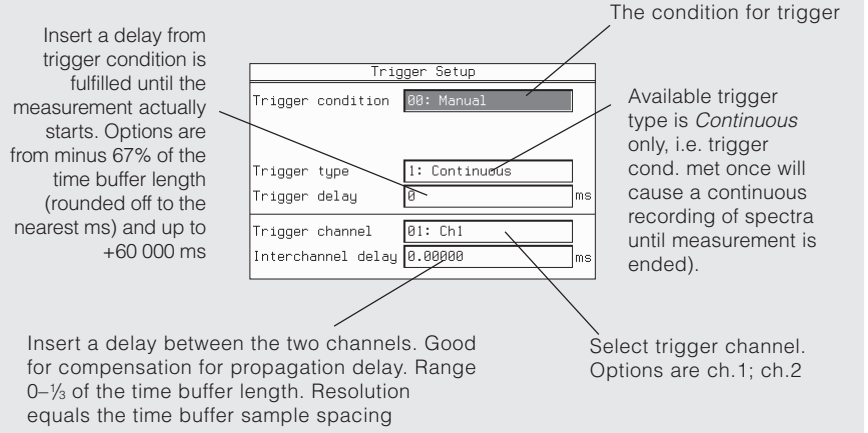
Likewise, if you select clock as trigger condition, extra fields will appear to let you define the time of trigger.

The trigger condition menu is accessed by pressing 2ND START.

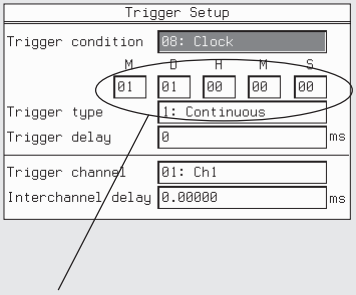
The six trigger conditions available in the FFT and how they work can be seen on the right part of this page spread.

The Menu for this Task

2nd Start

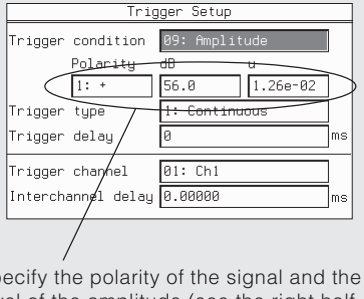

The condition for trigger
Insert a delay from trigger condition is fulfilled until the measurement actually starts. Options are from minus 67% of the time buffer length (rounded off to the nearest ms) and up to +60 000 ms
Trigger Setup
Trigger condition 00: Manual
Trigger type 1: Continuous
Trigger delay 0 ms
Trigger channel 01: Ch1
Interchannel delay 0.00000 ms
Available trigger type is *Continuous* only, i.e. trigger cond. met once will cause a continuous recording of spectra until measurement is ended).
Insert a delay between the two channels. Good for compensation for propagation delay. Range 0–⅓ of the time buffer length. Resolution equals the time buffer sample spacing
Select trigger channel. Options are ch.1; ch.2

If you select Clock as trigger condition, extra parameter-fields will appear…


Trigger Setup
Trigger condition 08: Clock
M D H M S
01 01 00 00 00
Trigger type 1: Continuous
Trigger delay 0 ms
Trigger channel 01: Ch1
Interchannel delay 0.00000 ms

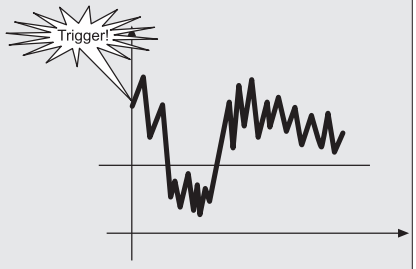
Specify time of trigger here as month, day of month and time of day

If you select Amplitude as trigger condition, extra parameter-fields will appear…



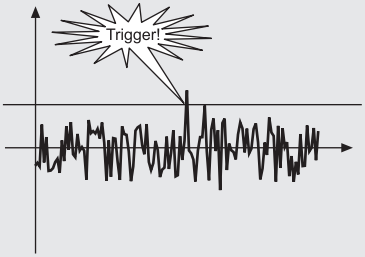
Specify the polarity of the signal and the level of the amplitude (see the right half of this page spread for more on this). You may specify in dB or units. If you set one, the other will be set to the corresponding value

Manual as trigger condition



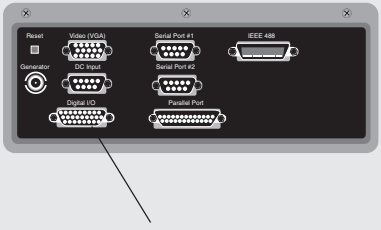
"as soon as START is pressed"

Amplitude as trigger condition



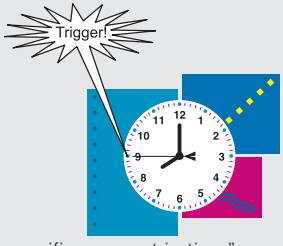
"specify amplitude level and polarity"

External... as trigger condition



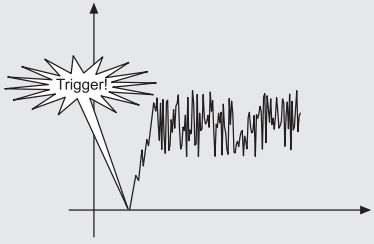
"grounding pin 23 on digital I/O will do it"

Clock... as trigger condition



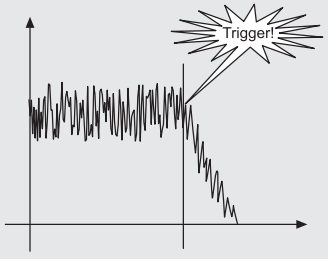
"at a specific moment in time"

Noise on... as trigger condition



"when you switch on the internal generator"

Noise off... as trigger condition



"when you switch off the internal generator"

Note: It may sound strange that a time signal's amplitude could be expressed in decibels. After all time signals are occasionally negative (sinusoids for instance tend to be negative for half the period) and the logarithm of negative figures fails to exist.

However, the analyser employs the same technique as the one used in Intensity mode, viz. to split the negative and the positive part of the signal and then treat them separately. In this way the decibel term retains its meaningfulness.

FFT mode supports single spectrum measurements only – no multi spectrum measurements can be made with FFT.

The Nor-840 makes all measurements with 67% overlap, i.e. all measurements made with Hanning weighting will be in real time – in dual channel mode and all the way up to 25 kHz!

The measurement setup menu is accessed by pressing the M.SETUP key.

What is Zero pad?

The FFT process assumes that the time buffer contains one period of a periodic signal. When computing correlation functions, these artificial periodic signals are displaced relative to each other and a circular correlation effect will appear. This is because the end of one signal period will overlap the beginning of the next period. The estimated correlation function will thus be incorrect.

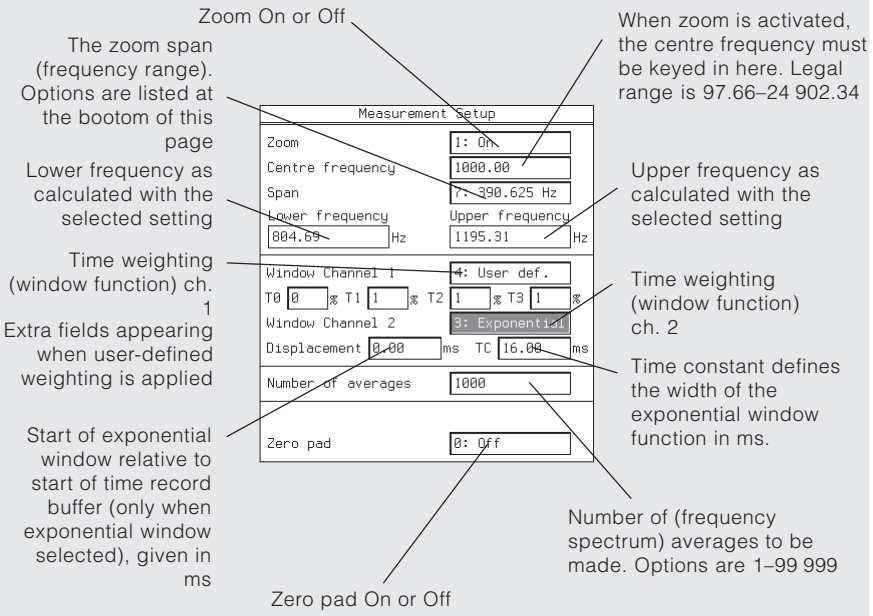
This can easily be overcome by setting the second half of the time buffer to zero. By doing this we introduce another error, but this error is known and can be compensated for (which is done, of course).

There are eight predefined frequency spans available for zooming

	Span	Δf
1:	25 000.0000 Hz	31.2500 Hz
2:	12 500.0000 Hz	15.6250 Hz
3:	6 250.0000 Hz	7.8125 Hz
4:	3 125.0000 Hz	3.9063 Hz
5:	1 562.5000 Hz	1.9531 Hz
6:	781.2500 Hz	0.9766 Hz
7:	390.6250 Hz	0.4883 Hz
6:	195.3125 Hz	0.2441 Hz

The Menu for this Task

M.Setup


Zoom On or Off
The zoom span (frequency range). Options are listed at the bootom of this page
Lower frequency as calculated with the selected setting
Time weighting (window function) ch. 1
Extra fields appearing when user-defined weighting is applied
Start of exponential window relative to start of time record buffer (only when exponential window selected), given in ms
Zero pad On or Off
When zoom is activated, the centre frequency must be keyed in here. Legal range is 97.66–24 902.34
Upper frequency as calculated with the selected setting
Time weighting (window function) ch. 2
Time constant defines the width of the exponential window function in ms.
Number of (frequency spectrum) averages to be made. Options are 1–99 999
Measurement Setup
Zoom
1: On
Centre frequency
1000.00
Span
7: 390.625 Hz
Lower frequency
804.69
Hz
Upper frequency
1195.31
Hz
Window Channel 1
4: User def.
T0 0 % T1 1 % T2 1 % T3 1 %
Window Channel 2
3: Exponential
Displacement 0.00 ms TC 16.00 ms
Number of averages
1000
Zero pad
0: Off

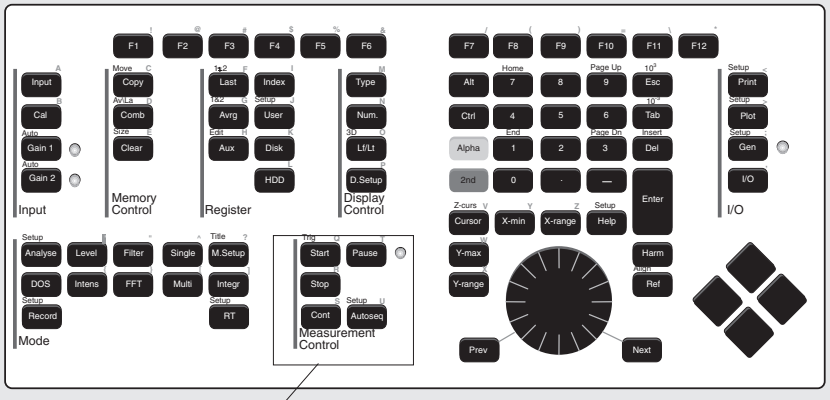
Tip: The effect the selected window function has on the time signal, as well as the look of the selected window function itself, can be displayed (but you need to make one measurement first) – see *The FFT Display Setup Menu* for details.

Tip: If the two channels are not set up with identical window functions, the setting of ch.2 will appear in reverse video in the display – see *A Tour of the FFT Mode Display* for details.

Would you like to read more?

A brief discussion of zoom and FFT windows functions is presented in the beginning of this section of the manual. An exhaustive presentation can be found in: Bendat J.S. and Piersol A.G. "Engineering applications of correlation and spectral analysis". John Wiley and Sons Inc. 1980 ISBN 0-471-05887-4

The Tool for this Task...



The measurement control keys

Tip: Once you enter the FFT mode you will note that the display is continuously updated. How fast an update depends on the complexity of the function displayed. For auto spectra the refresh rate is about 10 times per second. Once a started measurement is terminated, the display will "freeze" to show the data acquired during the measurement. The data acquisition is still active, however, running in the background but data are discarded and not retained. To return to the the *acquisition-show-and-then-discard* process press CLEAR LAST or change any of the measurement parameters.

Pressing:	Causes:
Start	the measurement to begin as soon as the trigger condition is met.
Pause	an ongoing measurement to be temporarily halted
Stop	an ongoing measurement to be terminated
Cont	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the preset number of averages is reached
Cont after the measurement end condition is met	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the preset number of averages is reached. The total number of averages made will be the sum of the two measurements'

Once the Nor-840 has been set up to your requirements, it is ready to make measurements.

To begin measuring, press the START key. The data acquisition will start as soon as the trigger condition is met. If you have set up a trigger delay, the acquisition will not begin until *a) the trigger condition has been met* and *b) any delay subsequently elapsed*.

The measurement will – if left uninterrupted – go on until the preset number of averages has been reached.

You may halt the measurement temporarily and then resume the measurement later. You may also stop the measurement prematurely by pressing the STOP key. The measurement will go on until the time buffer has been filled up and the corresponding spectrum calculated.

The instrument does not discriminate between pausing and termination with respect to resuming a halted measurement. Therefore, to pause the instrument just press the PAUSE or the STOP key.

To resume measuring, press the CONT key.

If the measurement was halted prematurely (paused), pressing CONT will cause the instrument to resume the measurement and go on until the preset number of averages has been reached.

On the other hand, if the CONT key is pressed after the preset number of averages has been reached, the measurement will be prolonged by a another set of averages equal to the preset number of averages.

In the latter case the total number of averages will be the sum of the two measurements (provided the measurement was not terminated prematurely during the prolongation).

A Tour of the FFT Mode Display

There are four display modes in the Nor-840 – two single function and two dual function modes.

When we say function display rather than channel display, this is because in dual function display mode, the two display halves are completely independent!

The upper display may show what you just measured, while the lower may show a graph of a stored measurement (or vice versa), or the upper display may show a graph while the lower shows a tabulation of the very same data etc.

When you select any of the two dual function displays (with or without setup information), the screen will consist of an upper and a lower part – which we may call windows – plus, whenever applicable the instrument and measurement setup information.

Only one window is active – i.e. responds to cursor movements etc. – at the time. The active window has a thick frame surrounding it.

Use the vertical pointing arrows of the ❖ (FIELD SELECTOR) keys to go between the windows.

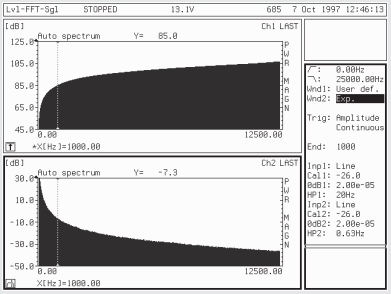
In each window you may set such display parameters as horizontal and vertical graduation, the vertical top scale and the horizontal datum (the lower end of the scale) as well move the graph cursor.

Observe that the alterations possible in the display have no effect on the measured or recalled data as such, only the appearance of the data will be affected.

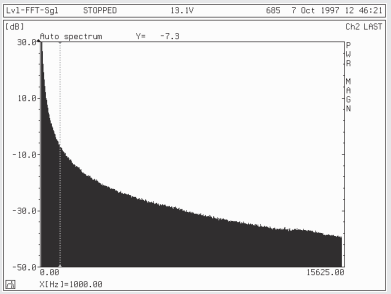
Use TYPE to switch between the display modes.

The four display modes…

Dual display with setup information

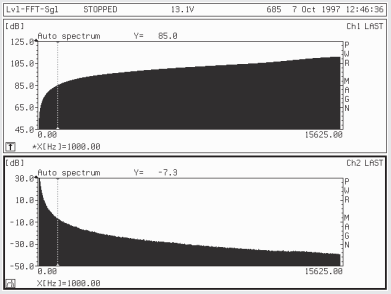


Single display (active window)

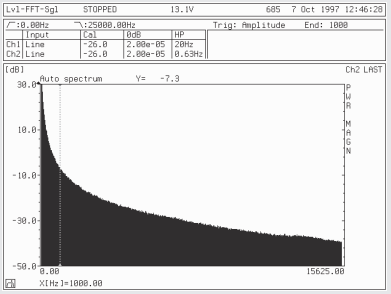


Type

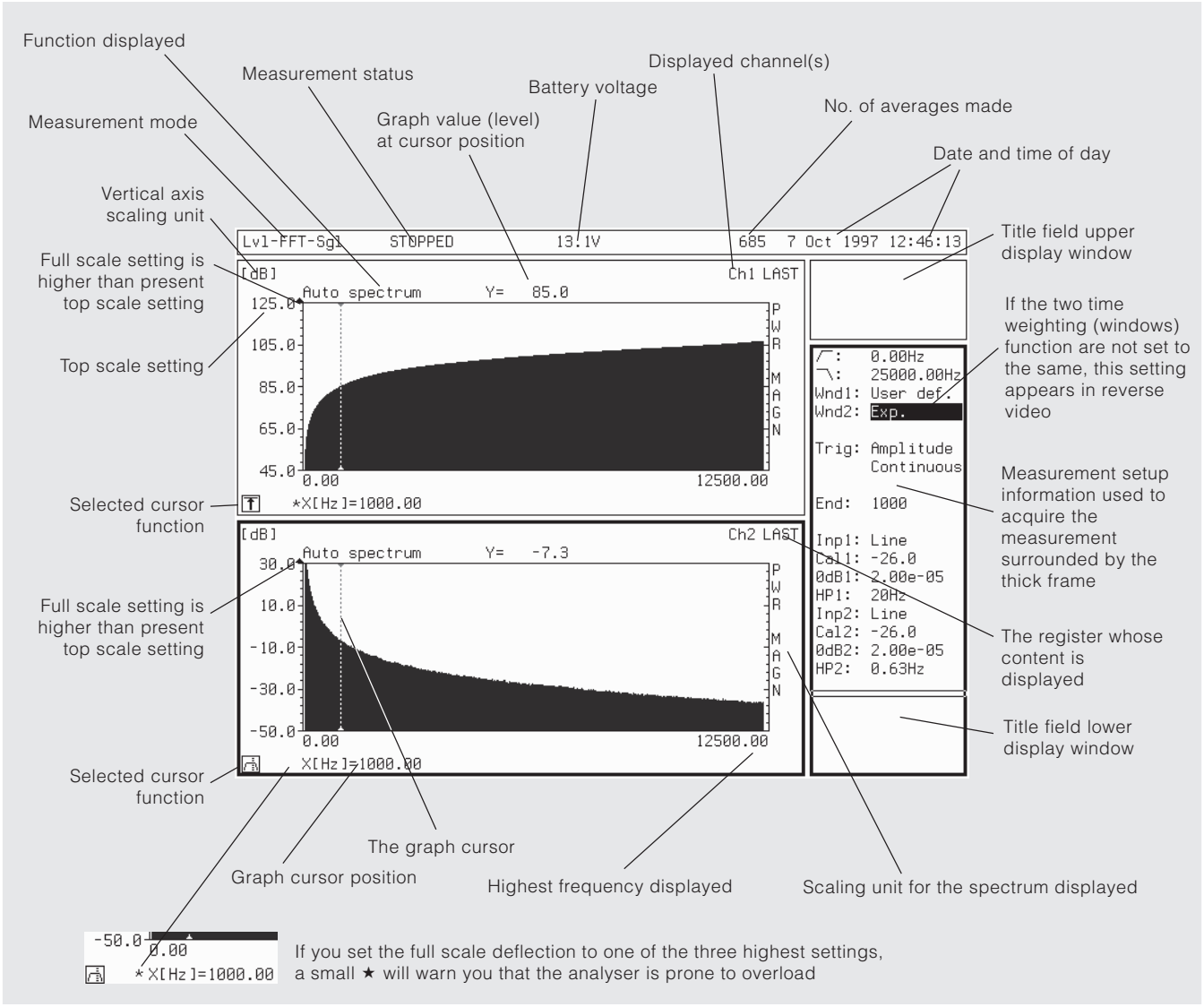
Dual display without setup information



Single display (active window) with condensed setup information



Note: The data of the active window will be the only set of data printed out when you make a numerical printout.


Function displayed
Measurement status
Battery voltage
Displayed channel(s)
No. of averages made
Measurement mode
Graph value (level) at cursor position
Date and time of day
Vertical axis scaling unit
Lvl-FFT-Sgl STOPPED 13.1V 685 7 Oct 1997 12:46:13
Title field upper display window
Full scale setting is higher than present top scale setting
[dB] Ch1 LAST
Auto spectrum Y= 85.0
125.0
105.0
85.0
65.0
45.0
0.00
12500.00
PWR MAG
Top scale setting
If the two time weighting (windows) function are not set to the same, this setting appears in reverse video
⌜: 0.00Hz
⌝: 25000.00Hz
Wnd1: User def.
Wnd2: Exp.
Trig: Amplitude
Continuous
End: 1000
Selected cursor function
*X[Hz]=1000.00
Measurement setup information used to acquire the measurement surrounded by the thick frame
[dB] Ch2 LAST
Auto spectrum Y= -7.3
30.0
10.0
-10.0
-30.0
-50.0
0.00
12500.00
Inp1: Line
Cal1: -26.0
0dB1: 2.00e-05
HP1: 20Hz
Inp2: Line
Cal2: -26.0
0dB2: 2.00e-05
HP2: 0.63Hz
Full scale setting is higher than present top scale setting
The register whose content is displayed
Title field lower display window
Selected cursor function
X[Hz]=1000.00
The graph cursor
Graph cursor position
Highest frequency displayed
Scaling unit for the spectrum displayed
-50.0
0.00
*X[Hz]=1000.00
If you set the full scale deflection to one of the three highest settings, a small ★ will warn you that the analyser is prone to overload

The FFT Mode Display Setup Menu

This setup menu controls how the acquired data are to be displayed.

There is one setup menu for each display window. The two are completely independent of each other. You can, however, copy the display setup for one of the windows to the other by the click of a button (the same way as you do in fractional octave band mode).

Pres D.SETUP to produce the display setup menu.

A Few Words on Functions

Some of the functions available are based on single channel measurements while others require dual channel measurements to become meaningful.

The two auto spectra and the corresponding auto correlations work with single channel measurements; all the others require dual channel measurements to work.

The correlation functions require zero pad activated and the weighted time function and the weighting window both require one measurement to be made first; and that the number of averages is set to no more than 1!

Displaying the Time Window Function

The Nor-840 offers you the option of displaying the selected time window function and the effect this has on the contents of the time signal buffer.

This, however requires that you make one measurement first and that you set the number of averages to 1. Since the analyser does not support time synchroneous averaging ("enhanced time") this is required to make the time buffer contents meaningful.

A description of how to display the time window function is given at the lower right part of this page spread.

Optimum Scaling of the Spectrum

The range of window functions available in the Nor-840 FFT extension gives the analyser the ability to analyse a wide variety of signal types.

The multitude of degrees of freedom that an FFT analyser provides generates many pitfalls which must be avoided. The shape of the window function and the frequency span of the measurement determine the noise bandwidth of the filters and the analysis time required. Consequently, you must apply correct units when scaling the frequency spectra acquired.

To be able to determine what will be the optimum scaling of a given spectrum, you must first evaluate what type of signal you are measuring on.

The signals can, in general, be divided into three types; *deterministic, random* and *transient*. A fundamental difference here will be the duration of the signal – the *determinstic* and the *random* signal types are both continuous while the *transient* type of signal is not.

A transient signal is – in this part of the manual – considerd to be a signal which starts and ends at zero amplitude within the time buffer length. The complete signal should be analysed in units of energy. Stationary continuous signals, on the other hand, should not be measured in terms of energy since the amount of energy measured will be directly proportional to the measurement duration. Therefore, power – which is energy per time unit – is a better choice.

Continuous signals are either random or deterministic. The former type will have a continuous spectrum, while the latter has a a spectrum consisting of lines (a line spectrum).

A continuous spectrum should be analysed in terms of spectral density – i.e. the measured level should be divided by the frequency unit (Hz that is). This is because the measured level will be proportional to the filter analysis bandwidth (at a relatively flat part of the spectrum).

The measured line spectra amplitude will – provided the the analysis bandwidth is sufficiently narrow to separate the individual components – be independent of the filter bandwidth.

The Nor-840 detects the RMS value of the signal. The FFT extension offers the following scalings:

Root Mean Sqare: RMS = √PWR; Power: PWR = RMS^2; Power Spectral Density: PSD = PWR/Bandwidth and Energy Spectral Density: ESD = PSD × Observation time.

We recommend the following:

- Scale in RMS or PWR when analysing periodic, determinstic signals
- Scale in PSD when analysing stationary random signals
- Scale in ESD when analysing transients.
- Sometimes signals are a combination of periodic and random (e.g sinusoids in noise), the RMS or PWR should then be used for scaling the sinusoidal components (the lines in the spectrum) and PSD for the continuous part of the spectrum, since this is originating from a random type of signal.

The Menu for this Task

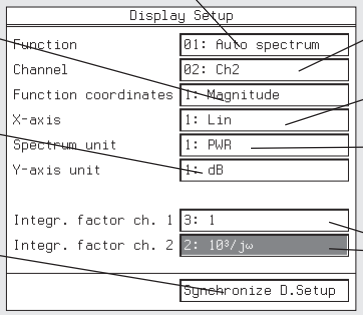
D.Setup

Function to be displayed. Options are: *01: Auto spectrum; 02: Cross spectrum; 03: Coherence; 04: Frq. response H1; 05: Frq. response H2; 06: Weighted time; 07: Weighting window; 08: Auto corr.; 09: Cross corr.*

Function coordinates
Options are: *1: Magnitude; 2: Phase; 3: Real part; 4: Imaginary part*

Vertical axis scaling unit.
Options are:
1: dB; 2: Units – Lin;
3: Units – log

Make the other display setup menu similar to this



Which channel? Options are: *Ch.1; Ch.2*

X-axis setting.
Options are *1: Line(ear); 2: Log(arithmic)*

Spectrum scaling unit.
Options are: 1: *PWR; 2: RMS; 3: PSD; 4: ESD*. See text for details.

Integration factor.
Options are: *1: $(10^3/j\omega)^2$; 2: $10^3/j\omega$; 3: 1; 4: $j\omega/10^3$; 5: $(10^3/j\omega)^2$*

Need to Flatten the Spectrum? Meet the Integration Factor

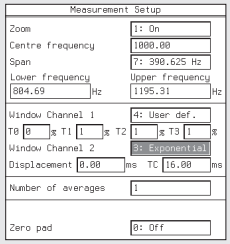
The display setup menu enables you to perform an integration or differentiation of the FFT spectrum. Applications for this feature include conversion from acceleration to velocity and displacement as well as vice versa, but also to make the spectrum fit better within the display range.

Note the gain/attenuation factor of 10^3, which has been included to make the processed graph fit within the display range. Double integration will then have an integration factor of 10^6! Do not forget to take this into account when you evaluate the results.

The feature is a display feature only. It has no effect on the measured data whatsoever!

To display the selected windows function or the time function weighted with the selected windows function…

❶

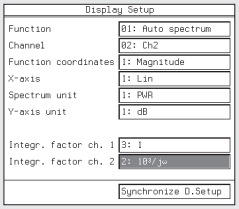


In the measurement setup menu, set the windows functions as required and the number of averages to 1.

❷

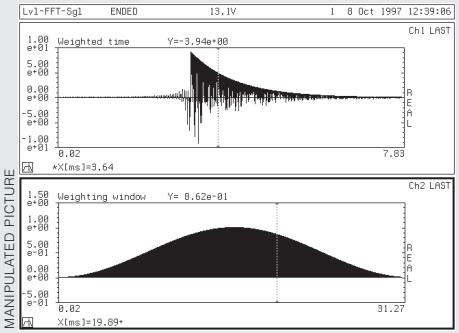
Press the START key

❸



In the display setup menu, select Weighting window or Weighted time as displayed function

❹



Example showing Weighted time (top) and Weighting window (bottom)

The FFT Mode Display Cursors

The Nor-840 has an extensive set of cursor functions. However, some apply to certain situations only.

All the cursor controls are located around the DIAL. To operate a cursor you start by selecting the type of cursor and then use the DIAL or PREV & NEXT. A small icon appears in the display to tell you which cursor function has been activated.

Scaling and Graduation

To optimise the presentation of the measured functions and values, you may adjust the horizontal and vertical axes' graduations (X-RANGE and Y-RANGE, respectively); the vertical axis top scale value (Y-MAX) and the horizontal axis minimum value (X-MIN).

Beam Finder – Locating the Graph

Sometimes the mismatch between the top scale setting and the values of the graph is so significant that no graph is shown in the display. What to do?

Enter the beam finder. A term borrowed from the world of oscilloscope technology where it was often difficult to tell which way to adjust the top scale to get the graph within the display.

If you press 2ND Y-MAX, the top scale value is changed by the analyser so that the graph appears in the display. The Y-MAX and Y-RANGE may then be used to refine the view as usual. In particular when using the integration/differentiation features of the display setup menu your graph may vanish from the screen – although we have tried to compensate somewhat for this by applying a multiplication factor of 10^3 to this function.

Cursor Alignment

In dual display mode the graph cursors (X-axis cursors) of the two windows may be aligned with each other and moved together in either direction.

Press 2ND REF to align cursors. To deactivate press again, press TYPE or change to another measurement mode (e.g. Level or Intensity).

The master cursor will normally be the cursor of the active window (the one surrounded by the thick frame).

However, if the frequency axis of one of the window's charts is set to logarithmic scaling while the other isn't, the chart set to logarithmic frequency axis will always become master.

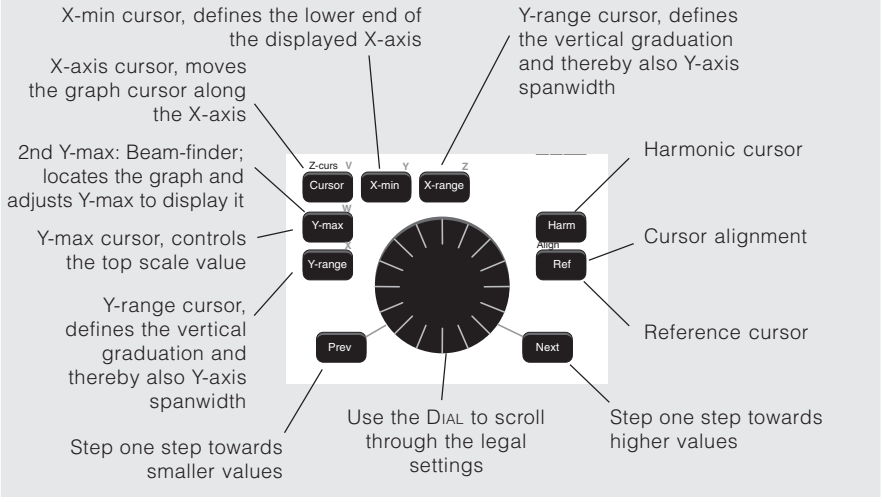
If you try to force it to behave otherwise, it will either make the logarithmic window active (first attempt) or switch the alignment off (second attempt).

If the frequency range of the master's graph exceeds that of the slave's, the slave's will stop at its end, while the master's cursor will go on when moving the cursor along the frequency axis.

If the frequency range of the master's graph is smaller than that of the slave's, the cursor will refuse to move outside the master's frequency range.

The slave's cursor will not move if the master's cursor is positioned at a line not present in the slave's graph. However, once the master's cursor is moved to a frequency line present in both graphs the slave's cursor will move immediately to that line.

The Cursor Control Keys


X-min cursor, defines the lower end of the displayed X-axis
Y-range cursor, defines the vertical graduation and thereby also Y-axis spanwidth
X-axis cursor, moves the graph cursor along the X-axis
2nd Y-max: Beam-finder; locates the graph and adjusts Y-max to display it
Y-max cursor, controls the top scale value
Y-range cursor, defines the vertical graduation and thereby also Y-axis spanwidth
Step one step towards smaller values
Use the DIAL to scroll through the legal settings
Step one step towards higher values
Reference cursor
Cursor alignment
Harmonic cursor
Z-curs
Cursor
X-min
X-range
Y-max
Y-range
Harm
Align
Ref
Prev
Next

Time functions can be aligned with time functions only

Reference Cursor

The reference cursor is used to investigate the difference between two points on a graph.

To activate the reference cursor, select CURSOR to move the graph cursor to your reference point, then press REF and use the DIAL or PREV & NEXT to move the graph cursor to a point elsewhere on the graph. The distance vertically and horizontally between the two points will now be shown in the active window.

Harmonic Cursor

The harmonic cursor is used to indicate any possible harmonic relationships present in the recorded spectrum.

The harmonic cursor is activated by pressing the HARM key.

The harmonic cursor will not appear whenever the graph cursor (frequency cursor) is located close to the lowest frequency in a baseband (unzoomed) measurement. The reason is simply that all you'll get when activating the harmonic cursor from this position is a screen covered with cursors.

Scaling and Graduation Range…

Y-axis: 20, 40, 60, 80 or 100 dB across the vertical scale or in a 1-2-5 sequence when set to engineering units.

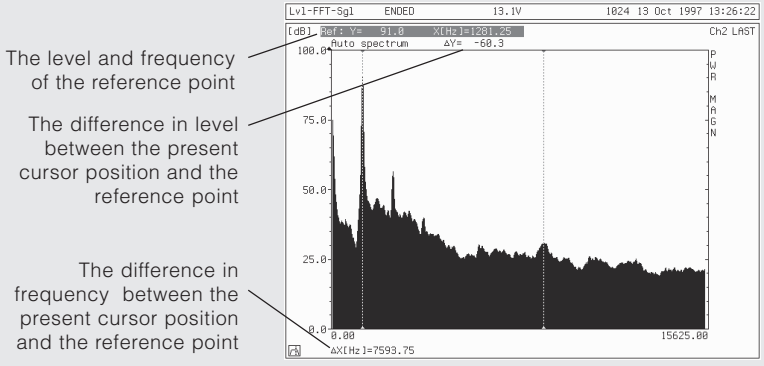
X-axis: 1:8, 1:4, 1:2, 1:1, 2:1, 4:1 and 8:1

The Icons Show the Type of Cursor Selected

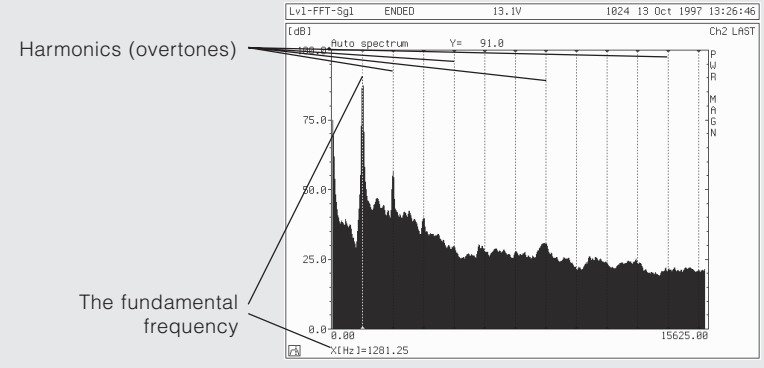
A B C D E

A: X-min
B: X-range
C: Y-max
D: Y-range
E: Graph cursor

The Reference Cursor



The Harmonic Cursor



Tip: Do not confuse top scale and full scale! The top scale is purely a display control indicating the top of the vertical axis, while the full scale controls the input amplifier gain settings. Changing the latter will delete the Last register contents!

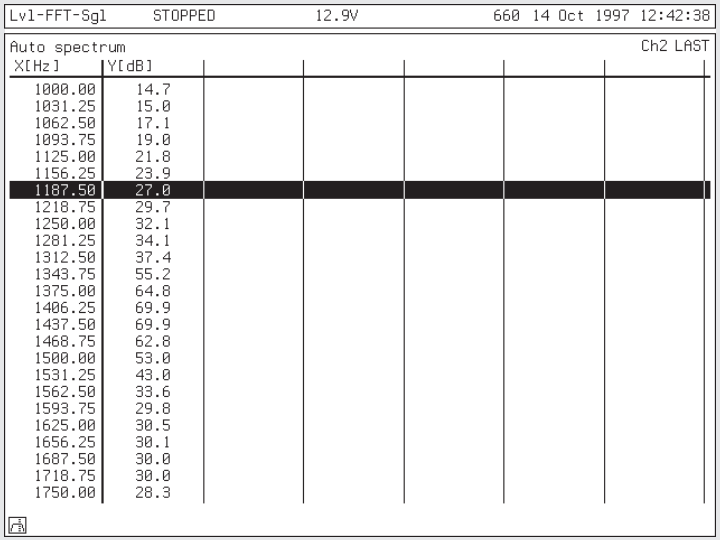
Data measured or retrieved from the disk may also be shown tabulated.

To switch between graphic and numerical presentation, use the NUM key.

The table is merely a numeric representation of the data.

The Numerical Table Is Merely a Numeric Representation of the Data

Num.

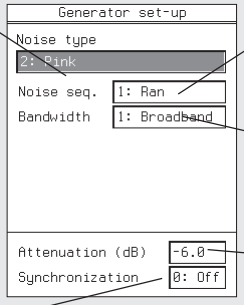


The Menu for this Task

2nd Gen

The noise type is
1:White
2:Pink
3:BP-filtered
4:Impulse
5: BP-filtered impulse
8: Multi-sine

Lets the generator span the measurement. When set to 1:On and START is pressed the noise generator will be turned on before the measurement is started and switched off after the measurement has ended

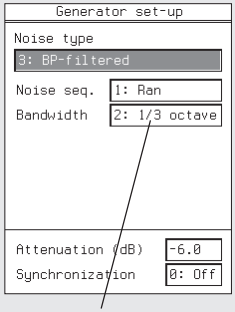


The noise sequence can be either *1:Ran*(dom) or *2:P*(seudo)*Ra*(ndom)

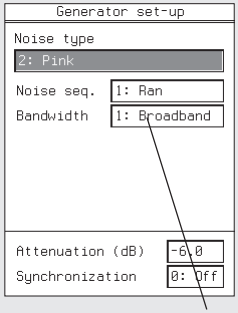
Extra field to define bandwidth when applicable – see below

Set the noise generator output level in dB re: 1V

Range: –40.0 dB to 0.0 dB in 0.1 dB steps



If you select bandpass filtered noise, you must specify the filter bandwidth (1/1 or 1/3 octave bands)



If you select pink noise, you must specify whether broadband (20–20 000 Hz) or limited (100–5 000 Hz) shall be used

This icon appears in the status line of the display when the noise generator is running

The Nor-840 comes with a powerful noise generator built in.

To access the setup menu press 2ND GEN.

The generator can be set up to supply white noise – whose level will increase by 3 dB per octave when viewed using fractional octave filters (but it looks flat when viewed using FFT because of its linear frequency axis); pink noise which looks flat in the fractional octave domain (but with a level decreasing by 3 dB per frequency doubling when viewed using FFT); bandpass filtered (1/1 or 1/3 octave bands only); impulse (1/1 or 1/3 octave bands only) and bandpass filtered impulse (1/1 or 1/3 octave bands only).

For the bandpass filtered signals, the frequency (band) is determined by the current graph cursor position.

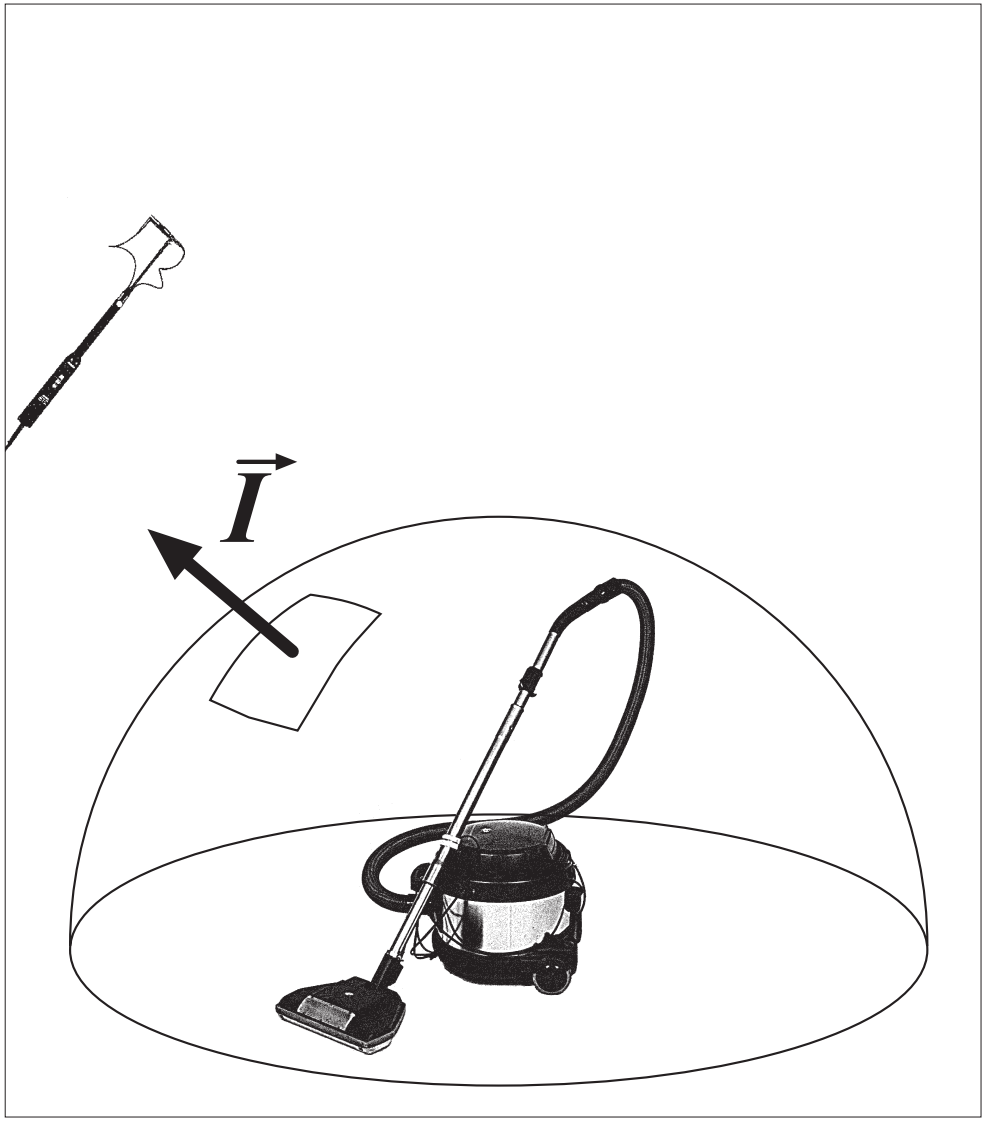
Once the generator has been properly set up and the setup menu closed, press GEN to activate the generator and again to deactivate.

What Is Multi-sine?

Multi-sine is a complex signal consisting of sinusoids. The frequency components of this signal coincides with the lines of the FFT spectrum. Hence there is no signal energy outside these lines. Such a signal can be shown to have a periodicity equal to the time buffer length. This means that rectangular weighting can be applied without introducing discontinuities at the extremes.

In addition the phase of each of the sinusoids has been adjusted to give a crest factor much lower than that of conventional noise – approximately 1.3 (a single sinusoid has a crest factor of 1.4142, i.e. √2 or the same as 3 dB).

Chapter 5



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Making Intensity Measurements

Intensity measurements are useful for the study of many sound propagation and transmission phenomena. Simultaneous observation of sound pressure and sound particle velocity can be used to determine the magnitude and direction of the energy propagation.

Sound intensity is a *vector* quantity defined as the average rate of flow of acoustic energy through a unit area, perpendicular to the direction of the wave propagation.

With mathematics:

$$\vec{I} = \frac{1}{T}\int_0^T p(t)\vec{u}(t)dt \quad [\mathrm{W/m^2}]$$

in which:

$p(t)$ is the instantaneous pressure and $\vec{u}(t)$ is the instantaneous particle velocity of the sound field.

Sound pressure is a *scalar* quantity having magnitude only, while the sound particle velocity is a vector quantity having both magnitude and direction.

The magnitude of the intensity can be measured in one particular direction, for example the *x*-direction, in which case:

$$I_X = \frac{1}{T}\int_0^T p(t)u_X(t)dt$$

in which the u_X denotes the instantaneous sound particle velocity in the *x*-direction.

The subscript *X* of I_X is normally not written, so the intensity component is denoted *I* only. In the rest of this manual, the symbol *I* should be interpreted as the intensity level along the axis of the sound intensity probe, unless explicitly stated otherwise.

From the above follows that a sound intensity probe must be able to measure the particle velocity as well as the sound pressure of the sound field.

The sound intensity level is referred to the value

$$I_0 = 10^{-12}\ [\mathrm{W/m^2}]$$

The equivalent continuous sound intensity level can then be expressed as

$$I_{eq} = 10\log\frac{|I|}{|I_0|}\ [\mathrm{dB}]$$

Sound power may be calculated directly from the intensity values provided that you have information on the area over which the intensity was measured.

The sound power may then be expressed as

$$L_W = I_{eq} + 10\log S\ [\mathrm{dB}]$$

in which *S* is the area in $\mathrm{m^2}$.

The following annotation is used throughout this section of the manual:

SIL: Sound Intensity Level
I_{eq}: Equvivalent Continuous Sound Intensity Level
L_W: Sound Power Level

Direction is indicated as follows:

(+)–23.6 equals a positive intensity of –23.6 dB re. 10^{-12} $[\mathrm{W/m^2}]$

Sound Intensity Probes

Traditionally, two types of sound intensity probes used to be around, viz. the *p-p probe* and the *p-u probe*.

The *p-p probe* (the *two-microphone probe)* uses the sound pressure from two closely spaced microphones to calculate the sound particle velocity by applying Newton's second law (mass × acceleration = force):

$$\rho \frac{\partial \vec{u}}{\partial t} = -gradp$$

also called *Euler's Relation,* in which ρ is the density of the air and $\vec{u}$ is the particle velocity.

In one direction, x we have

$$\rho \frac{\partial u_x}{\partial t} = -\frac{\partial p}{\partial x}$$

Since the pressure gradient is proportional to the particle acceleration, the particle velocity can be obtained by integrating the pressure gradient with respect to time.

$$u_x = -\frac{1}{\rho}\int \frac{\partial p}{\partial x} dt$$

For practical cases, the pressure gradient is approximated by measuring the sound pressures p_1 and p_2 at two closely spaced points and dividing the pressure difference $p_1 - p_2$ by the microphone separation distance Δx.

The estimate for the particle velocity $\hat{u}_x$ in the x-direction will then be

$$\hat{u}_x = -\frac{1}{\rho \Delta x}\int (p_2 - p_1) dt$$

This is an approximation, but it is valid for separation distances much smaller than the examined wavelengths of the sound field.

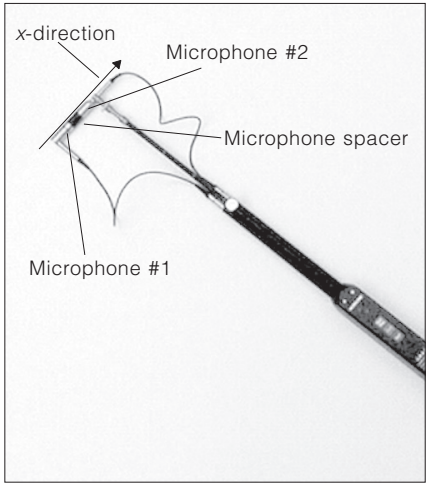
A practical probe can therefore be designed using two closely spaced microphones. From the two measured sound pressures the mean sound pressure and the particle velocity (from the pressure gradient) are calculated in the direction of a line joining the two microphones' centres – see Fig below. In this way we obtain both the magnitude and the direction of the sound intensity.

The *p-u probe* on the other hand is based on a somewhat different principle. As a practical example we will take a look at the now discontinued intensity probe Nor-216 which consisted of two sets of transducers – a microphone for the sound pressure and a sound velocity transducer for detecting the particle velocity directly.

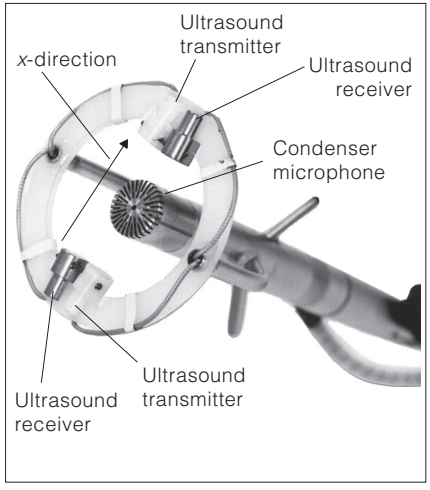
The velocity transducer utilised the interaction between an ultrasonic wave and the audio sound field that we want to measure. The ultrasonic wave will travel faster in an airflow in the same direction as the sound propagation and slower in an airflow of the opposite direction.

Two pairs of ultrasonic transmitters and receivers were placed in antiparalel close to each other. The particle velocity of the sound field then acted as an oscillating airflow with the direction changing in accordance with the frequency of the sound field. By comparing the phases of the two received ultrasound signals the difference inthe transmission velocity was determined. This was a direct indicator for the sound particle velocity.

A p-p probe example – the Nor-240…



A p-u probe example – the Nor-216…



The Nor-840 has four different types of signal input sockets. These are accessed via the input menu. Press the INPUT key to produce this menu.

Only one input can be selected at the time per channel, but the two channels need not be set to the same type of input source.

This opens up for *non-mainstream* intensity measurements, for example intensity measurements using a microphone and an accelerometer.

Observe that when you select sound intensity for one of the channels, the other channel will be set to the same mode automatically, since sound intensity is a two-channel measurement technique. If you then change the setting of one of the channels, the other will go back to the setting it had at the time intensity was selected.

A lowpass filter will be activated when *3: Charge* is selected.

There is a highpass filter for each channel. If you activate this, it will affect the setting of the lowest frequency band to be measured.

The highpass filters are laser trimmed to maintain maximum phase accuracy. Together with the choice quality components used this ensures a stable phase response throughout the temperature range your Nor-840 has been designed for.

This setting is found in the measurement setup menu.

The new setting will be the first centre frequency above that of the highpass setting provided the initial setting was lower than or equal to what the highpass filter is set to. If it was set to a higher value initially, it will remain unaffected.

Although the input selection is a global setting (affecting all modes), only the current mode (single spectrum or multispectrum) will be affected by the forced change of lower end frequency range setting. Otherwise measured data not stored would have to be deleted to avoid inconsistencies.

However, this means that if you make a measurement in any other mode without modifying the lower end setting of the frequency range, the frequency bands in the vicinity of the highpass filter cutoff frequency will be biased due to the presence of the highpass filter.

In the current mode, you may change the setting of the lowest frequency band to be measured, to the value of your liking *after* the highpass filter has been activated.

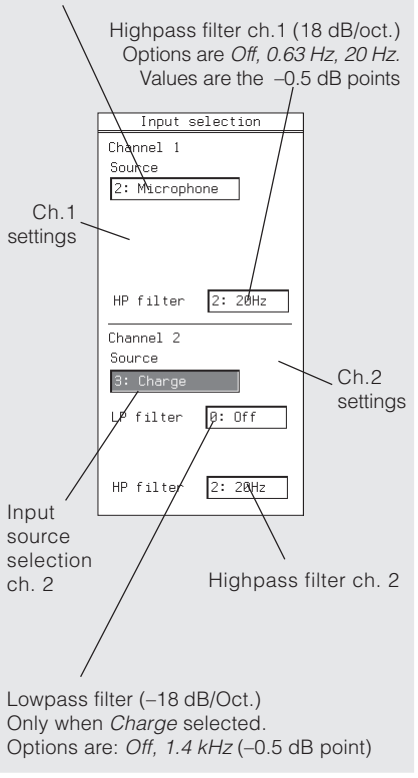
The change is made inside the measurement setup menu for the current mode.

Should you again modify the highpass filter setting and set it to a higher value (in Hz), the same will take place again provided the lower end setting is lower or equal to the highpass filter cutoff frequency. Otherwise it will be left unaffected.

If you decide to modify the lower end keep in mind that the above biasing of the lower frequency bands will apply to the current mode as well.

The highpass filter cutoff frequency stated is the –0.5 dB frequency point (not the usual –3 dB frequency point).

The Menu for this Task Input



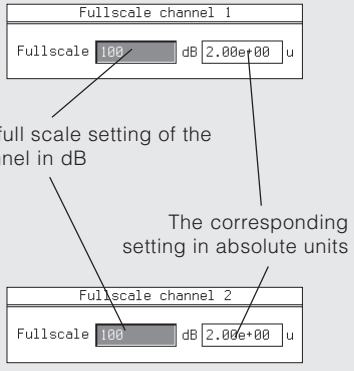
Tip: Always set the two highpass filters to the same cut-off frequency. By doing so you ensure that any interchannel phase-mismatch is without significance with respect to intensity measurements.

Note: This is not the place where you enter intensity mode – that is done by pressing INTENSITY, FILTER, SINGLE or INTENSITY, FILTER, MULTI. Probe selection (p-p or p-u probe) is done in the measurement setup menu

The Menu for this Task

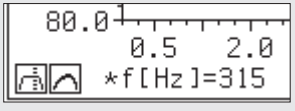
Gain1
Gain2

There is one menu per channel


Fullscale channel 1
Fullscale 100 dB 2.00e+00 u
The full scale setting of the channel in dB
The corresponding setting in absolute units
Fullscale channel 2
Fullscale 100 dB 2.00e+00 u

Do not confuse the full scale setting and the full display setting! The former defines the input amplifier gain and hence the overload margin for a given signal level, while the latter is used to set the display to make the measured signal fit within the setting of the axis.

If you set the full scale setting to one of the three highest settings, a small * will warn you that the analyser is prone to overload...



The available full scale setting of a measurement is given as a combination of the input amplifier gain setting and the calibration setting.

The gain is set separately for each channel and is adjustable in steps of 5 dB.

Press GAIN1 (GAIN2) to enter the menu.

The full scale setting controls the setting of the input amplifiers. However, it has no influence on the vertical scale in the display. The two extremes (top and bottom) of the vertical scale is controlled by the *top scale value*, which has its own dedicated key called the Y-MAX key. The top scale setting is purely a display function having nothing to do with the input amplifier whatsoever.

Note that if you set the full scale setting to a very high value, you face the risk of having severe overload in the transducers without seeing any trace of it in the analyser – simply because the analyser input isn't overloaded. For example; a ½" microphone with a sensitivity of 50mV/Pa will distort severely (more than 3% total harmonic distortion) when exposed to levels above 146 dB(A). To warn you about this the three highest settings provide an asterisk (*) in the display.

Note: Autoranging is not available in any of the sound intensity modes.

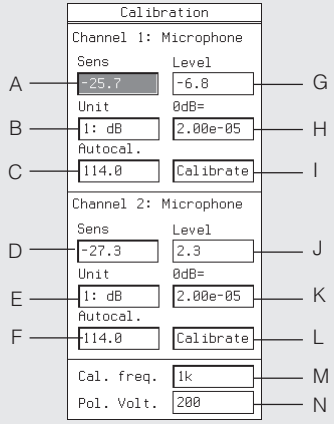
Intensity calibration should be carried out in Level mode, calibrating each part of the transducer separately. This applies to both p-p and p-u probes. You may want to check the system for residual intensity. Turn page for a description of this.

Although you may calibrate by just keying in the sensitivity, we always recommend that you use a calibrator. This is the only way to ensure proper operation of the entire measuring chain including the transducer(s).

If you are going to make non-mainstream intensity measurements (e.g. involving one microphone and one accelerometer) it may be convenient to change the 0 dB level of the relevant channel to obtain dB readings easy to compare with other vibration measurements. Check with relevant Standards and conventions to find suitable or commonly used 0 dB levels.

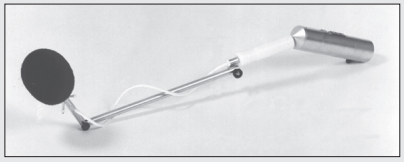
Note that if you set the 0 dB level to a value different from 2×10^{-5}, the selected setting will appear in reverse video on the screen as shown in the lower Fig. to the right.

The Menu for this Task Cal



A Sensitivity of channel 1
B Ch. 1 units (dB or engineering units)
C Auto-calibration level of channel 1
D Sensitivity of channel 2
E Ch. 2 units (dB or engineering units)
F Auto-calibration level of channel 2
G Level of ch. 1 measured with the selected time constant
H 0 dB level of ch. 1
I Initiate auto-calibration of ch. 1
J Level of ch. 2 measured with the selected time constant
K 0 dB level of ch. 2
L Initiate auto-calibration of ch. 2
M Calibrator frequency
N Polarisation voltage setting

When Using the Nor-216



Intensity measurements may also be made with the now discontinued p-u probe Nor-216.

The cable Nor-1460 must be used to connect the probe to the analyser.

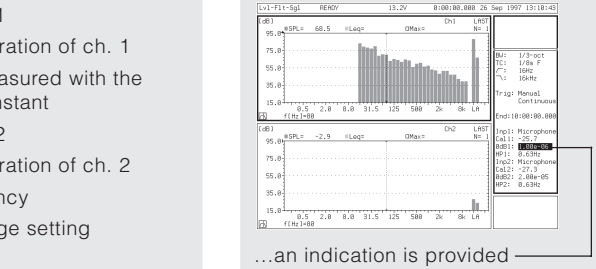
Analysers with serial number 17857 and lower must undergo a minor hardware modification to be able to support the use of Nor-216.

The probe type selection is made in the measurement setup menu.

The sensitivity of the velocity channel must be set 52 dB above the sensitivity that was used when the probe was used with the Nor-830.

This means that you should key in +26 dB as the velocity sensitivity setting and not –26 dB as when used with the Nor-830.

If you Change the 0 dB Reference level



…an indication is provided

Calibration Using a Sound Calibrator

❶

Insert microphone into calibrator

❷

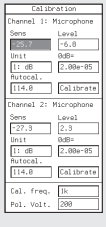
Press INPUT and select input source

❸

Adjust the full scale setting (GAIN1/GAIN2)

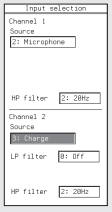
❹

Press CAL and set the calibrator frequency

❺ 

Set the sensitivity of the microphone used until the sound pressure level is indicated correctly

Calibration, Setting the Sensitivity

❶ 

Press INPUT and select input source

❷

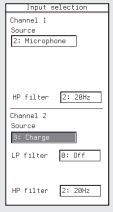
Press CAL and set the sensitivity of the microphone used

This method is in general not recommendable since it does not take into account whether the microphone actually works as it is supposed to do. Bearing in mind that the microphone itself is the most vulnerable part of the measurement chain, this method obviously suffers from severe shortcomings.

Auto-calibration Using a Sound Calibrator

❶

Insert microphone into calibrator

❷ 

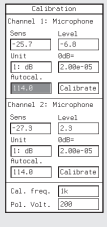
Press INPUT and select input source

❸

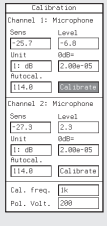
Adjust the full scale setting (GAIN1/GAIN2)

❹

Press CAL and set the calibrator frequency

❺ 

Set the calibrator's output level

❻ 

Select *Calibrate* and press ENTER to commence auto calibration

Checking Residual Intensity Using the Nor-1254 Sound Intensity Calibrator

After you have calibrated your sound intensity system, you may want to check for residual intensity.

By residual intensity we mean virtual intensity appearing when both transducers (microphones in most cases) of a p-p probe is exposed to the same sound field.

Any residual intensity present stems mainly from interchannel phase mismatch.

To do this you will need an intensity calibrator – such as the Nor-1254 – to be able to expose the two microphones to the same sound field simultaneously. This calibrator is available separately.

The Nor-1254 is a two port calibration coupler for phase and level calibration of sound intensity pairs. When used with the Nor-840, the calibrator is connected to the noise generator output of the analyser.

The sound intensity calibrator Nor-1254 consists of a sound source supplying a well defined sound pressure field simultaneously to the diaphragms of the two measurement microphones of the p-p probe. The microphones can be ½" or ¼", the latter by means of an adaptor.

The microphones are inserted into the holes of the Nor-1254 as shown in the photo below right. The noise generator of the Nor-840 is connected to the BNC socket.

For a sound pressure level of approximately 110 dB SPL the generator should supply approximately 300 mV$_{RMS}$. The electrical input signal to the Nor-1254 should never exceed 1 V$_{RMS}$.

The Nor-1254 can be used in two modes; viz. as a P-I index calibrator or as an intensity/particle velocity calibrator – cf. the user documentation for the Nor-1254.

In the P-I index mode the two microphones are exposed to exactly the same sound field so that the difference in phase response between the two can be determined.

In the intensity/particle velocity calibration mode the sound pressure signal fed to one of the microphones is phase shifted relative to the signal fed to the other. The phase shift is introduced by inserting an acoustic resistor in the coupler between the two microphones.

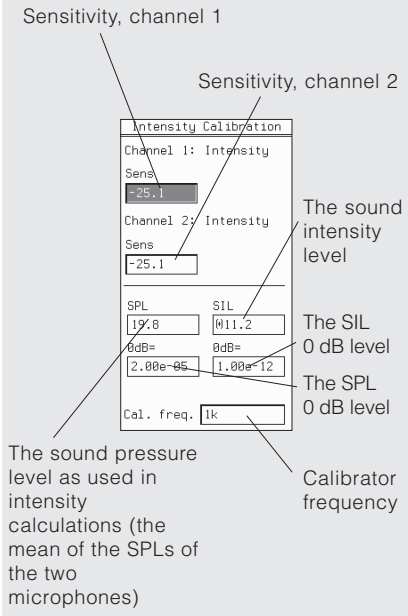
P-I Index Measurement Procedure

Before you attempt to measure the P-I index of your probe, you should check that the phase match of your analyser is sufficient. This is done by feeding the same electrical signal (from the noise generator of the analyser) to the two inputs and then read the difference between the sound pressure level and the sound intensity level – the P-I index.

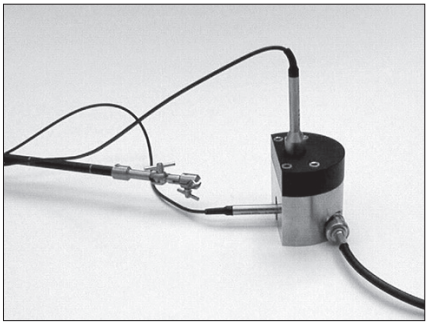
When it has been assured that the P-I index of the analyser is large enough, the intensity probe, consisting of the sound intensity microphone pair and preamplifiers can be checked with the Nor-1254.

The IEC 61043 Standard contains requirements to sound intensity measurement equipment. We recommend that you read through that Standard.

The Menu for this Task



The sound intensity calibrator Nor-1254…



Minimum P-I index [dB] requirements for 25 mm microphone separation (from IEC 61043-1993)

Band centre frequency Hz	Probe		Nor-840		Nor-840+Probe	
	Class1	Class2	Class1	Class2	Class1	Class2
50	13	7	19	13	12	6
60	14	8	20	14	13	7
80	15	9	21	15	14	8
100	16	10	22	16	15	9
125	17	11	23	17	16	10
160	18	12	24	18	17	11
200	19	13	25	19	18	12
250	20	14	26	20	19	13
315	20	15	26	20	19	14,5
400	20	16	26	20	19	15
500	20	17	26	20	19	16
630	20	18	26	20	19	16
800	20	18	26	20	19	16
1000	20	18	26	20	19	16
1250	20	18	26	20	19	16
1600	20	18	26	20	19	16
2000	20	18	26	20	19	16
2500	20	18	26	20	19	16
3150	20	18	26	20	19	16
4000	20	18	26	20	19	16
5000	20	18	26	20	19	16
6300	20	18	26	20	19	16

Note: We recommend an electrical P-I index of 30 dB or better when checking probe alone

Note: Long averaging times are necessary, e.g. to determine a P-I index of 20 dB @ 50 Hz the averaging time must be at least four minutes!

Tip: Reproducing measured values for intensity probe P-I indices greater than 25 (approximately) has proven to be very difficult. Very small phase changes are likely to produce significant variations in the P-I index.

Checking P-I Index for the Nor-840

1. Enter Intensity mode. Set spacer length to 25 mm in the measurement setup menu
2. Connect the noise generator to both line inputs. Be sure to set both inputs to Line
3. Select pink noise or red /white (if available)
4. Start a measurement
5. Use the display setup menu to set up the display to show the P-I index directly. For frequencies above 300 Hz we recommend a P-I index of at least 29 dB. Minimum requirements are listed in the table to the left.

Checking P-I Index for Probe+Nor-840

1. Mount the probe and enter intensity mode. Set 25 mm as spacer length in the measurement setup menu
2. Connect the noise generator to the Nor-1254. Set the input selectors of the Nor-840 to those used with your p-p probe
3. Select pink or red-white noise (if available) and adjust the output level to about –5 dB.
4. Set FSD so that the SPL level is about 10 dB below FSD to get an optimum signal-to–noise ratio
5. Start a measurement
6. Use the display setup menu to set up the display to show the P-I index directly. See the left table for minimum requirements.

The measurement setup menu is used to set up essential measurement parameters.

The menu comes in two flavours; one to cover the setup of single spectrum measurements and another to cover that of the multispectrum level measurement.

To access the menu press the M.SETUP key.

If you alter the high pass filter setting of the input selection menu, this will cause the lower setting on the frequency range to be changed if the initial setting was lower than that of the new high pass filter setting.

If it was set to a higher value initially, it will remain unaffected.

You may change the lower end setting of the frequency range in the measurement setup menu after having set the high pass filter in the input menu. However, if you set the frequency range to include frequency bands below the high pass filter setting, the readings of these frequency bands will be biased because of the high pass filter.

Note that changing the high pass filter setting affects the measurement setup menu of the selected mode (single spectrum or multispectrum) only. Other modes will not be affected (and may hence yield biased data) to avoid inconsistencies.

As long as the highpass filters of the two channels are set to the same cut-off frequency, their influence on residual intensity will be insignificant.

Multispectrum measurements consist of a series of consecutive measurements referred to as *periods*. All periods will have the same duration (specified in the measurement setup menu) and there is no loss of data in-between the periods.

You may set up your Nor-840 to capture the time profile of what goes on immediately before the trigger condition is met. This is done by specifying that a certain number of the multispectrum periods shall apply to the situation before trigger.

A more exhaustive explanation of this can be found in the article *The Level Mode Measurement Setup Menus* of the section *Making Level Measurements*, and in the illustrations to the extreme right of this page spread.

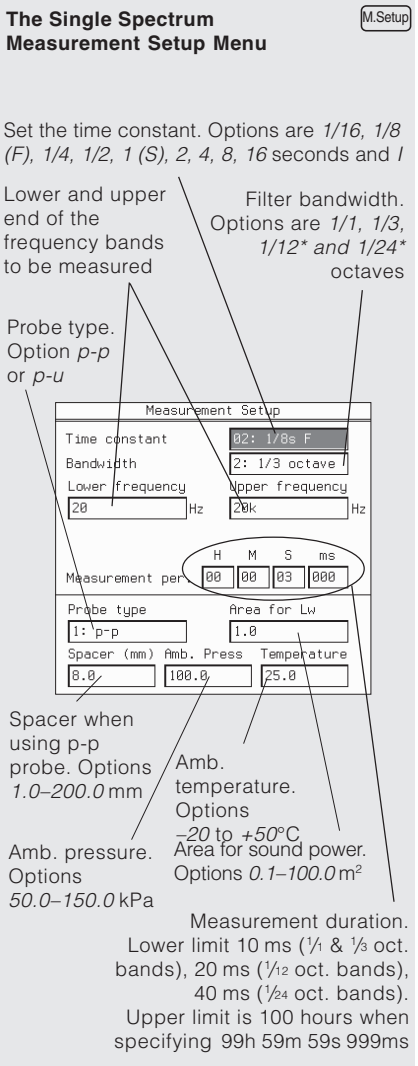
Specifying Area for Sound Power

The surface area for sound power calculations is set up in this menu. Normally, you would set up the area before you make your measurement. However, for convenience we have added the option of changing the area even after you made your measurement. This applies to the contents of Last register only.

To utilise this post-measurement feature, press the M.SETUP key after the measurement and change the Area for Lw. *Change nothing else!* Press ENTER to exit the menu, as usual.

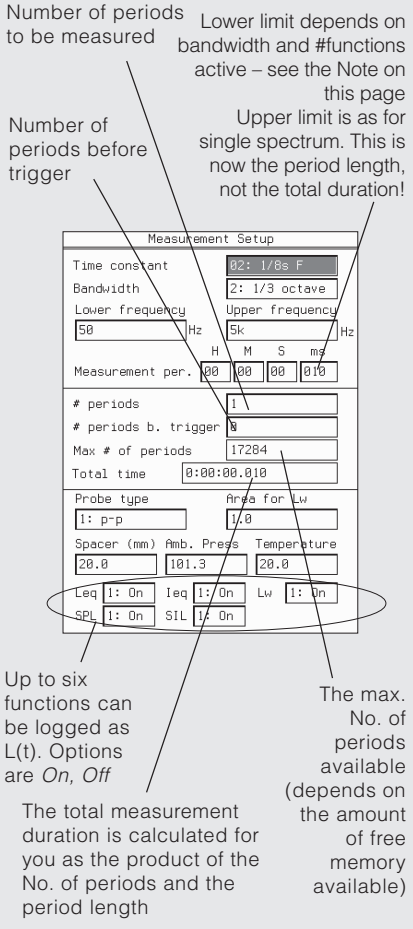
Based on the keyed-in area, the sound power is calculated for that particular surface. When you transfer these data to the Average register to average them with power data from other surfaces, the effect of the surface area is implicitly taken into account. Thus the use of surface parts with non-identical areas will represent no problem.

The Single Spectrum Measurement Setup Menu


M.Setup
Set the time constant. Options are 1/16, 1/8 (F), 1/4, 1/2, 1 (S), 2, 4, 8, 16 seconds and I
Lower and upper end of the frequency bands to be measured
Filter bandwidth. Options are 1/1, 1/3, 1/12* and 1/24* octaves
Probe type. Option p-p or p-u
Measurement Setup
Time constant
02: 1/8s F
Bandwidth
2: 1/3 octave
Lower frequency
20
Hz
Upper frequency
20k
Hz
H M S ms
Measurement per
00 00 03 000
Probe type
1: p-p
Area for Lw
1.0
Spacer (mm)
Amb. Press
Temperature
8.0
100.0
25.0
Spacer when using p-p probe. Options 1.0–200.0 mm
Amb. temperature. Options –20 to +50°C
Amb. pressure. Options 50.0–150.0 kPa
Area for sound power. Options 0.1–100.0 m²
Measurement duration. Lower limit 10 ms (1/1 & 1/3 oct. bands), 20 ms (1/12 oct. bands), 40 ms (1/24 oct. bands). Upper limit is 100 hours when specifying 99h 59m 59s 999ms

* Optional, must be ordered separately

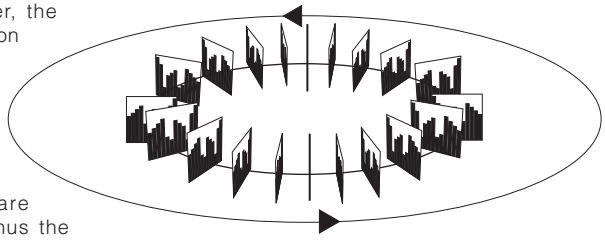
The Multispectrum Measurement Setup Menu is just an extension of the single spectrum setup menu…

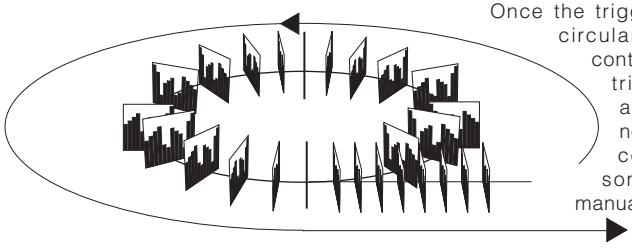


Functions and features not explained here is explained in the single spectrum setup menu (to the left).

In multispectrum mode the Nor-840 can be set up to record a number of spectra before trigger…

When you have set up the Nor-840 to record a part of the periods before trigger, the acquisition of data starts upon pressing the START key. The data are stored in a circular buffer whose length exactly matches the number of periods to be recorded before trigger. When the buffer gets full, the oldest data are automatically overwritten. Thus the buffer will always contain the latest periods acquired.





Once the trigger condition is met, the circular buffer will be retained containing the periods before trigger while the data acquisition will go on as normal. Note that the trigger condition must be set to something different from manual to make this feature work.

Note: The lower limit for the period length in multispectrum mode is bandwidth dependent; 4 ms (1/1 & 1/3 oct. bands), 10 ms (1/12 oct. bands) and 20 ms (1/24 oct. bands). If more than one function is set active, the lower limits are 10 ms (1/1 & 1/3 oct. bands), 20 ms (1/12 oct. bands) and 40 ms (1/24 oct. bands).

Tip: In multispectrum mode, the number of periods available depends on the amount of free memory available. However, if you can do with fewer functions logged simultaneously, or with a more narrow range of frequency bands, the free memory available can be spent on what you really need. The number of periods available will be doubled if the number of functions logged is halved. Likewise it will be doubled if the frequency range is halved.

Therefore, we recommend that you consider your needs before you set up the analyser. Is the time domain resolution you have chosen appropriate – or is it overkill, drowning you in data? If you can get away with, for instance, half the time resolution (i.e. doubling the period length, you will be able to cover the same total duration with only half the number of periods (so maybe you could keep that many functions activated after all?).

Sound Intensity Trigger Conditions – Fractional Octave Analysis

There are separate trigger setups for each of the two modes (one for the single spectrum mode and another for the multispectrum mode). In this way the two modes are kept apart.

If you want to record the time profile of what takes place immediately before the trigger condition is met, you may set up the analyser to do so. However, this applies to multispectrum measurements only and it is not done in the trigger setup menu, but in the (multispectrum) measurement setup menu.

The trigger condition menu is accessed by pressing 2ND START.

The nine trigger conditions available and how they work can be seen on the right part of this page spread.

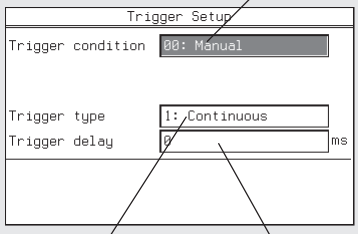
For p-p probes, the trigger condition you define will be applied to the mean sound pressure level, i.e. $\frac{1}{2}[p_1(t) + p_2(t)]$.

For all other probes or configurations, the trigger condition will be applied to the SPL of channel 1. The corresponding time constant is defined in the measurement setup menu.

The Menu for this Task

2nd Start

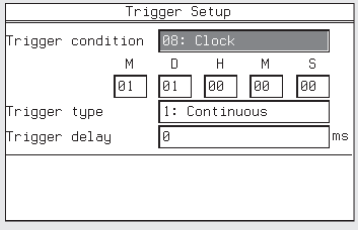
The condition for trigger



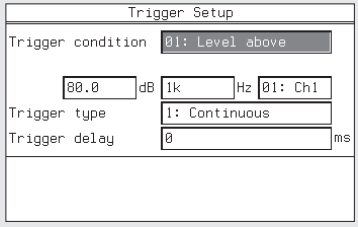
Insert a delay from trigger condition is fulfilled until the measurement actually starts. Options are 0–60 000ms

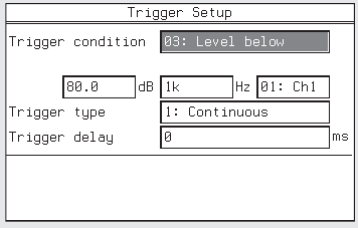
Not used in this software version

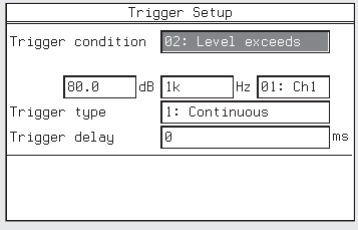
If you select Clock as trigger condition, extra parameter-fields will appear…

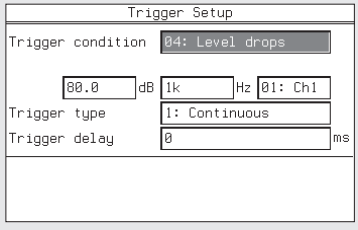


If you select level related trigger conditions, extra parameter-fields will appear…

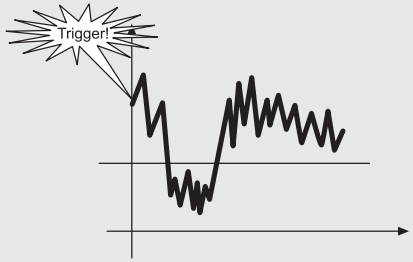






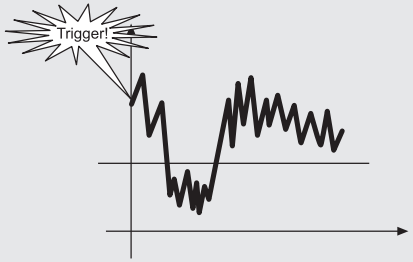


Manual as trigger condition



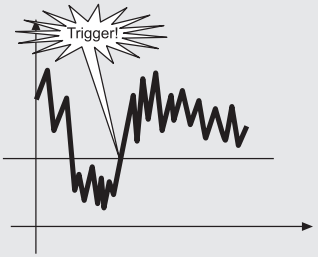
“as soon as START is pressed”

Level above… as trigger condition



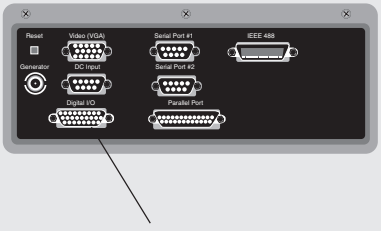
“whenever the level is above the threshold”

Level exceeds… as trigger condition



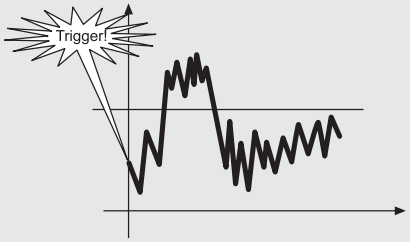
“transition is the keyword”

External… as trigger condition



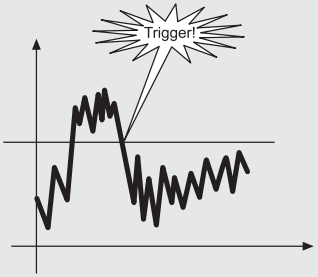
“grounding pin 23 on digital I/O will do it”

Level below… as trigger condition



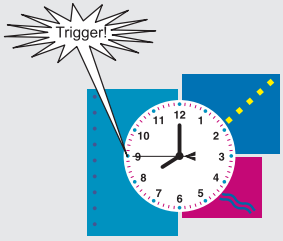
“whenever the level is below the threshold”

Level drops below… as trigger condition



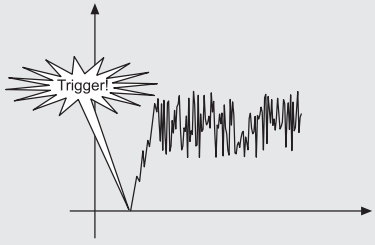
“transition is the keyword”

Clock… as trigger condition



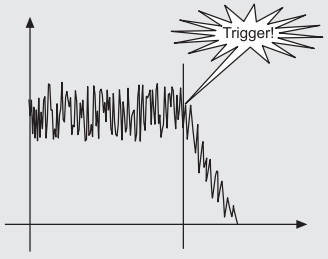
“at a specific moment in time”

Noise on… as trigger condition



“when you switch on the internal generator”

Noise off… as trigger condition



“when you switch off the internal generator”

Sound intensity measurements can also be made with FFT-technique as an alternative to the more conventional fractional octave method.

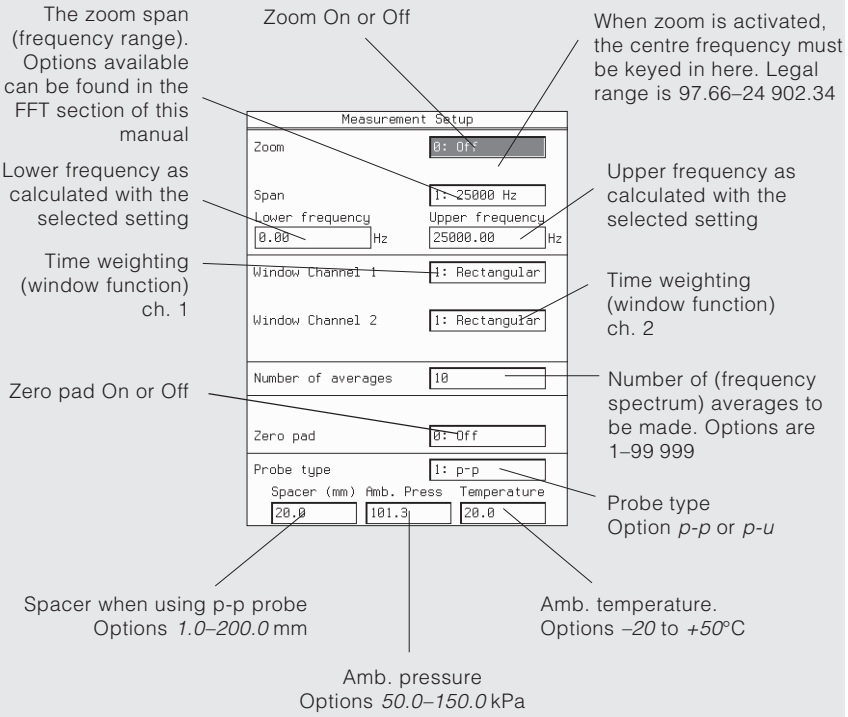
The measurement setup menu is identical to the FFT measurement setup menu, with the exception of two extra lines added to provide probe definition.

Automated sound power calculations cannot be made with FFT-based sound intensity.

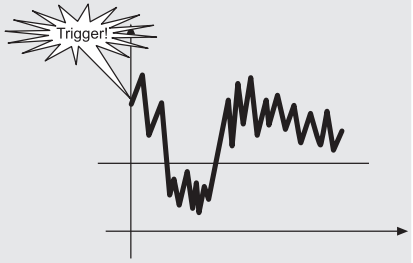
For details on aspects of setting up for FFT measurements not covered here, see the section *Making FFT Measurements*.

The Menu for this Task

M.Setup

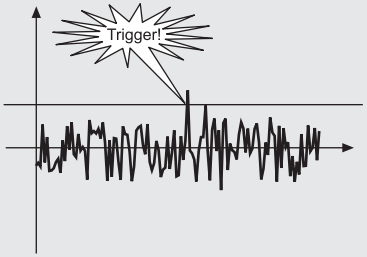


Manual as trigger condition



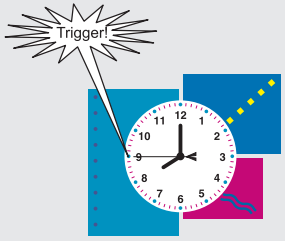
"as soon as START is pressed"

Amplitude as trigger condition



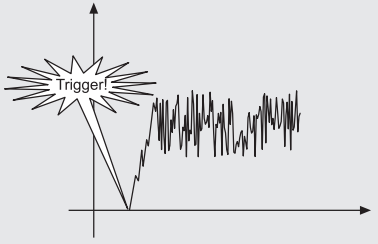
"specify amplitude level and polarity"

Clock… as trigger condition



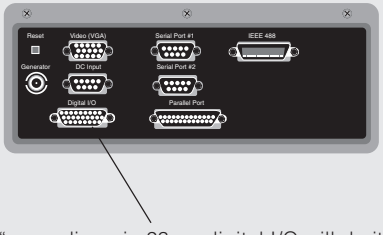
"at a specific moment in time"

Noise on… as trigger condition



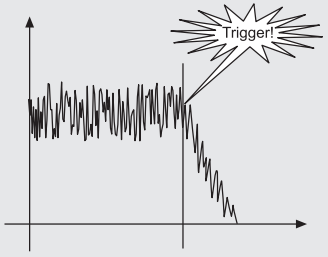
"when you switch on the internal generator"

External… as trigger condition



"grounding pin 23 on digital I/O will do it"

Noise off… as trigger condition



"when you switch off the internal generator"

The trigger conditions of FFT-based intensity are the same as those applying to other FFT mode measurements. In fact the FFT-based intensity should be regarded as a subset of FFT as such.

The FFT-based intensity trigger condition setup menu is identical to the fractional octave based trigger condition menu. However, for obvious reasons some of the trigger conditions available with fractional octaves become meaningless when working with FFT.

The trigger conditions shown here are those applying to FFT-based intensity.

The trigger condition menu is accessed by pressing 2ND START.

For a view of how the menu looks, turn to the article *Trigger Conditions in FFT Mode* of the section *Making FFT Measurements.*

Once the Nor-840 has been set up to your requirements, it is ready to make measurements.

To begin measuring, press the START key. The data acquisition will start as soon as the trigger condition is met. If you have set up a trigger delay, the acquisition will not begin until *a) the trigger condition has been met* and *b) the delay subsequently elapsed*.

The measurement will – if left uninterrupted – go on until the measurement end condition is met.

You may halt the measurement temporarily and then resume the measurement later. You may also stop the measurement prematurely by pressing the STOP key.

The instrument does not discriminate between pausing and termination with respect to resuming a halted measurement. Therefore, to pause the instrument just press the PAUSE or the STOP key.

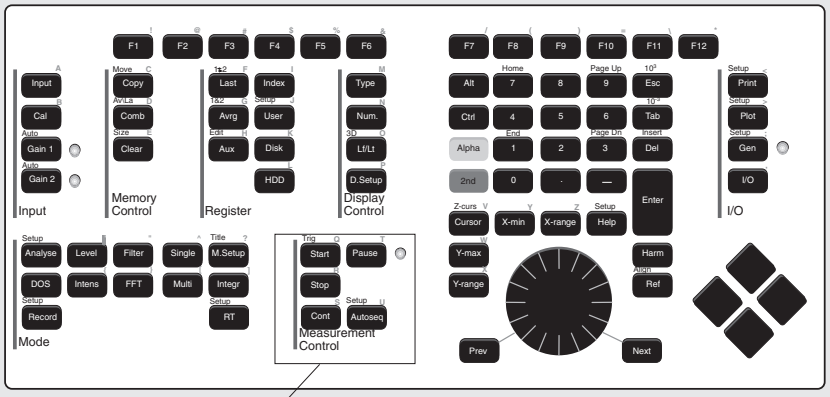
To resume measuring, press the CONT key.

If the measurement was halted prematurely (paused), pressing CONT will cause the instrument to resume the measurement and go on until the preset measurement duration expires.

On the other hand, if the CONT key is pressed after a measurement has ended successfully (i.e. the preset duration has expired) the measurement will be prolonged by a another period equal to the preset duration.

In the latter case the total duration will be the sum of the two durations (provided the measurement was not terminated prematurely during the prolongation).

The Tools for this Task

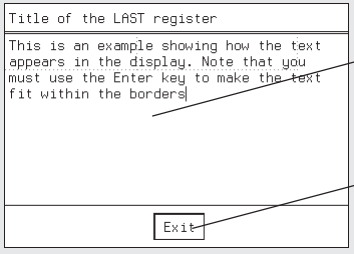


The measurement control keys

Pressing:	Causes:
Start	the measurement to begin as soon as the trigger condition is met.
Pause	an ongoing measurement to be temporarily halted
Stop	an ongoing measurement to be terminated
Cont	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the measurement end condition is met.
Cont after the measurement end condition is met	the measurement to be resumed. If left uninterrupted, the measurement will then go on until the measurement end condition is met. The total measurement duration will be the sum of the durations of the two measurements.

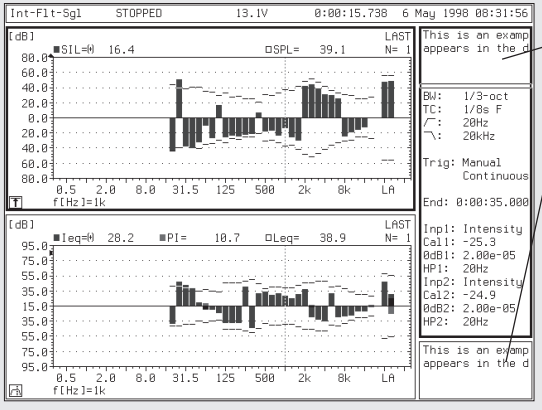
The Menu for this Task

2nd M.Setup


Title of the LAST register
This is an example showing how the text appears in the display. Note that you must use the Enter key to make the text fit within the borders
Exit

Enter the text here. The analyser will enter alpha mode upon entering this menu, so you won't have to press ALPHA first

To exit the menu, you must use the FIELD CURSOR to move to this field and then press ENTER



…and here is how the text appears in the display. Note how the text has been truncated. Only the text written inside the dotted field appears in the display when exiting the menu. The text is the same for the upper and lower window since they belong to the same measurement

You may assign a title to your measurement. This may help you keep track of what was measured and where.

The title is assigned to the contents of the current register of the current mode. If you move or copy the contents of the current register, a copy of the title follows the measured data.

If you assign a title to the Last register and let it remain unchanged, all following measurements will acquire the very same title, since the title is assigned to the contents of the Last register – whatever that may be.

We therefore recommend that you consider updating this title regularly to make it meaningful.

If you change to another measurement mode, the currently selected register of the new mode will acquire the same title as the same register of the previous mode, unless that register of the new mode had a title already and it was empty in beforehand.

To access the Title menu, press 2ND M.SETUP.

Tip: If automatic file name generation has been set to off and a title is present, the eight first characters of the title – ignoring any spaces – will be proposed as file name when storing the acquired data.

There are four display modes in the Nor-840 – two single function and two dual function modes.

When we say function display rather than channel display, this is because in dual function display mode, the two display halves are completely independent!

The upper display may show what you just measured, while the lower may show a graph of a stored measurement (or vice versa), or the upper display may show a graph while the lower shows a tabulation of the very same data etc.

When you select any of the two dual function displays (with or without setup information), the screen will consist of an upper and a lower part – which we may call windows – plus, whenever applicable the instrument and measurement setup information.

Only one window is active – i.e. responds to cursor movements etc. – at the time. The active window has a thick frame surrounding it.

Use the vertical pointing arrows of the ❖ (FIELD SELECTOR) keys to go between the windows.

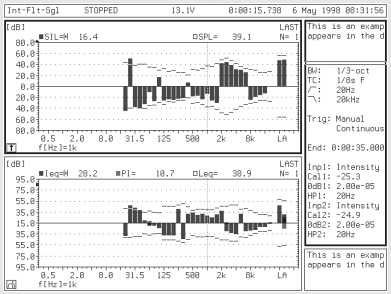
In each window you may set such display parameters as horizontal and vertical graduation, the vertical top scale and the horizontal datum (the lower end of the scale) as well as move the graph cursor.

Observe that the alterations possible in the display have no effect on the measured or recalled data as such, only the appearance of the data will be affected.

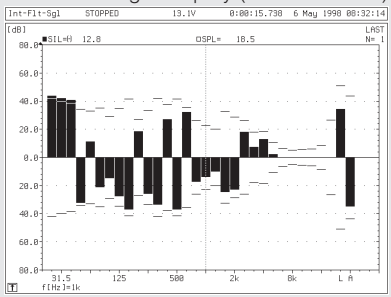
Use TYPE to switch between the display modes.

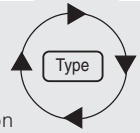
The Four Display Modes

Dual display with setup information

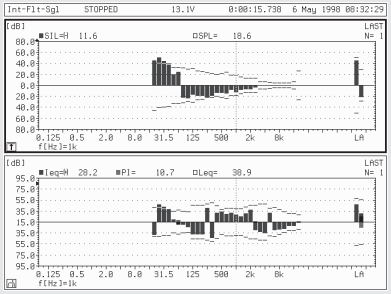


Single display (active window)

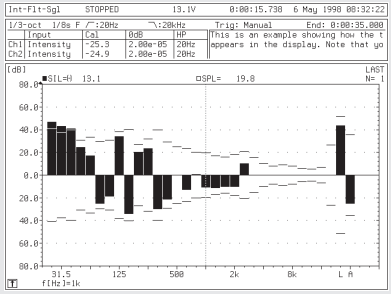




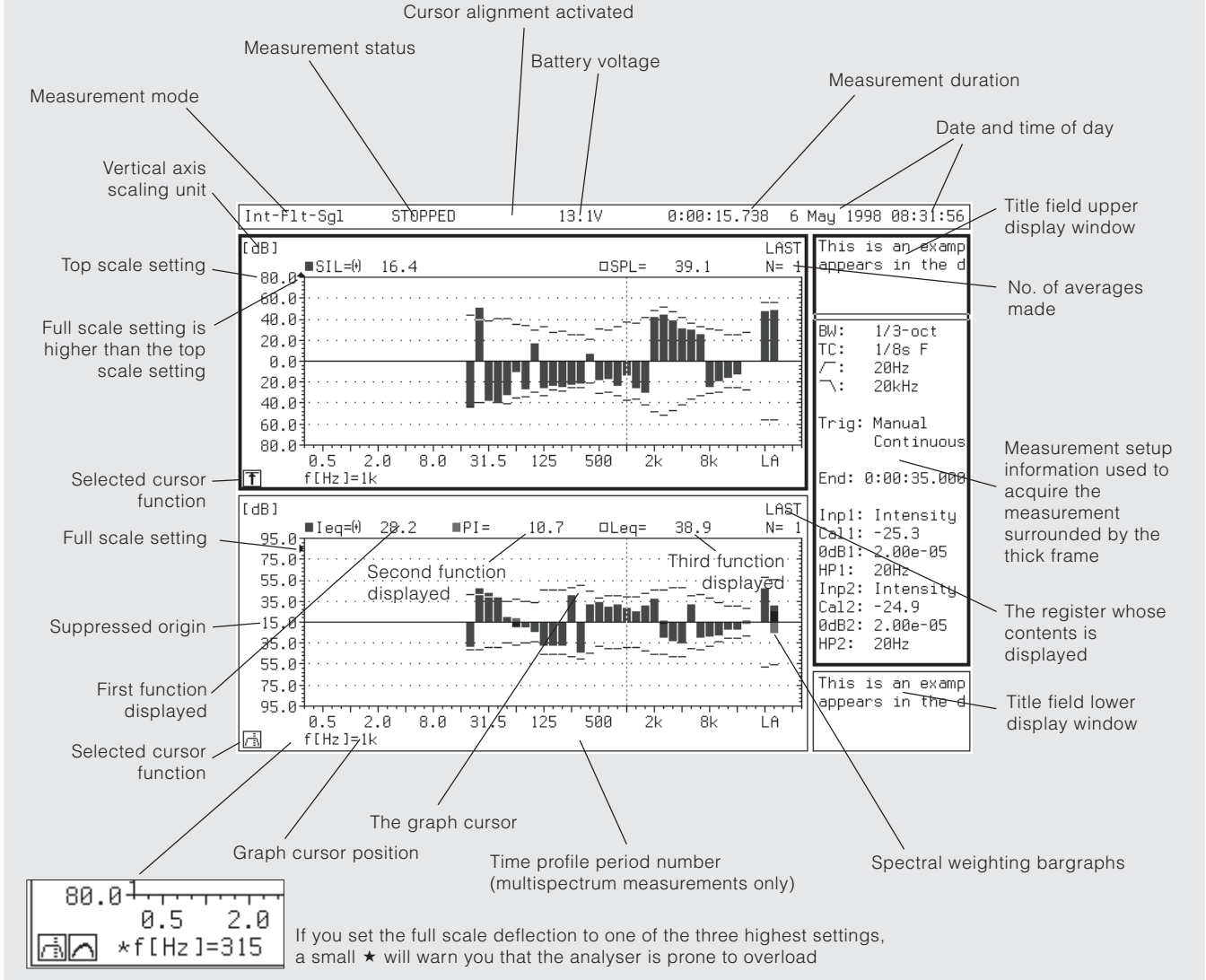
Dual display without setup information



Single display
with condensed setup information



Note: The data of the active window will be the only set of data printed out when you make a numerical printout.


Cursor alignment activated
Measurement status
Battery voltage
Measurement mode
Measurement duration
Date and time of day
Vertical axis scaling unit
Int-Flt-Sgl STOPPED 13.1V 0:00:15.738 6 May 1998 08:31:56
Title field upper display window
Top scale setting
[dB] LAST
SIL=(H) 16.4 SPL= 39.1 N= 1
This is an examp
appears in the d
No. of averages made
Full scale setting is higher than the top scale setting
80.0 60.0 40.0 20.0 0.0 20.0 40.0 60.0 80.0
0.5 2.0 8.0 31.5 125 500 2k 8k LA
f[Hz]=1k
BW: 1/3-oct
TC: 1/8s F
/‾: 20Hz
‾\: 20kHz
Trig: Manual
Continuous
End: 0:00:35.000
Inp1: Intensity
Cal1: -25.3
0dB1: 2.00e-05
HP1: 20Hz
Inp2: Intensity
Cal2: -24.9
0dB2: 2.00e-05
HP2: 20Hz
Measurement setup information used to acquire the measurement surrounded by the thick frame
Selected cursor function
[dB] LAST
Ieq=(H) 28.2 PI= 10.7 Leq= 38.9 N= 1
Full scale setting
95.0 75.0 55.0 35.0 15.0 35.0 55.0 75.0 95.0
Second function displayed
Third function displayed
Suppressed origin
The register whose contents is displayed
First function displayed
This is an examp
appears in the d
Title field lower display window
Selected cursor function
0.5 2.0 8.0 31.5 125 500 2k 8k LA
f[Hz]=1k
The graph cursor
Graph cursor position
Time profile period number (multispectrum measurements only)
Spectral weighting bargraphs
80.0
0.5 2.0
*f[Hz]=315
If you set the full scale deflection to one of the three highest settings, a small ★ will warn you that the analyser is prone to overload

This setup menu controls the type of functions (values measured, e.g. Ieq or Imax) to be displayed, the scaling of the axes and the layout of the numerical table.

There is one setup menu for each display window and one set of setup menus for the single spectrum mode and another for the multispectrum mode – i.e. one set of two menus for each mode.

These two sets of setup menus are completely independent of each other; while you can copy the display setup of the upper window to the lower window (and *vice versa*), you cannot copy between the single spectrum and the multispectrum modes.

Press D.SETUP to produce the Display Setup menu.

Displayed Curves

The sound intensity is shown on a bipolar basis. For convenience the origin will be suppressed when a combination of *Y-range* and *Y-max* does not allow the origin to be included. Similarly, a very low Y-max setting may set the vertical datum (the lower end of the scale) to a value below 0 dB.

You may also set the analyser to display sound pressure related functions such as SPL and Leq. Note that these will be shown symmetrically about the frequency axis to make comparisons between sound pressure and sound intensity easy to do.

For p-p probes, the sound pressure level is the mean sound pressure level, i.e. $\frac{1}{2}[p_1(t) + p_2(t)]$. For all other types of probes the SPL will be taken from channel 1.

The Numerical Table

The Display Setup menu is also used for setting up the functions to appear in the numerical table.

The table consists of six columns. If you set e.g. three columns to display the same function, this function will appear three times in the table.

In addition to the functions available for graphical display, the table also offers the option of displaying the number of averages made within each filter band (the average counter).

Tip: The A- and Linear spectral weighting networks are "true" spectral weighting functions in the sense that they are separate measurement filter bands. All other spectral weighting functions are applied as postprocessing features only.

This means that if you synthesise, e.g. a B-weighting curve and fail to apply a full bandwidth to your measurement (by measuring 50–3150Hz only for example), the B-weighted values will normally deviate from those obtained with a true spectral weighting filter band. The only way to make them similar is to apply your function to measurements using a full bandwidth.

Tip: When applying spectral weighting functions other than A- and Linear (see above) the term maximum values become meaningless since we have no information on when the maximum level occurred.

Tip: Numerical (tabulated) printouts will contain the functions set active in the numerical table part of the Display setup menu only. Make sure that the functions here are functions measured, otherwise your printout will contain one or more empty columns.

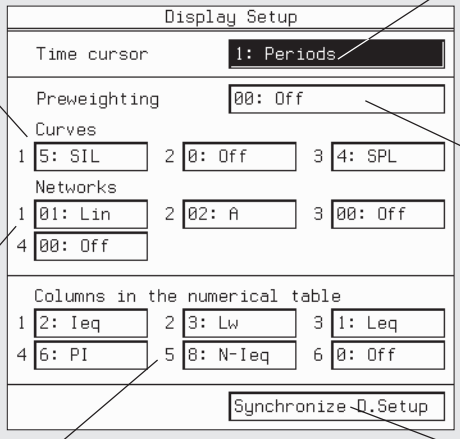
The Menu for this Task (there is one independent menu for each of the two display windows) D.Setup

Up to three set of curves (graphs) may be shown simultaneously. Which functions to show is defined here. Choose between *Off, Leq, Ieq, Lw, SPL, SIL, PI.*

In multispectrum mode, only the functions actually recorded will provide graphs

There are up to four bargraphs located to the right of the spectrum in the display. Select which ones to appear among: *A, Lin, SumA†, SumB, SumC, SumL* or *W1–W8**

The contents of the numerical table is determined by the setup of these parameter fields. Both the numerical display and the numerical output (printout) is affected by this setup. The options to select from are the same as those applying to the displayed curves plus N-Leq and N-Ieq



Time axis scaling. Applies to multispectrum measurements only. Select between *Periods*, *Relative time* (since trigger) and *Absolute time* (date and time of day)

The displayed graph can be shown spectrally weighted. Options are *A, B, C, L, W1–W8**

There is one display setup menu for each of the two windows (the upper and the lower window). Instead of going to the other window and set up the display, you can apply the same setup to the other window by moving the cursor to this field and then exit the menu (by pressing ENTER)

* W1–W8 denotes the spectral weighting function you can make yourself, either by keying in the gain/attenuation values of each frequency band or by converting a measurement to a weighting curve

† The term SumA denotes the A-weighted value calculated from the measured spectrum, while the A as such is a true A-weighting filter applied to the measurement as an independent measurement channel. The two may differ for two reasons; round-off differences in the digital calculation process (normally small deviations) and if the frequency range of the measured spectrum is limited – see the tip on the left side of this page spread.

The Nor-840 has an extensive set of cursor functions. However, some apply to certain situations only.

All the cursor controls are located around the DIAL. To operate a cursor you start by selecting the type of cursor and then use the DIAL or PREV & NEXT. A small icon appears in the display to tell you which cursor function has been activated.

Scaling and Graduation

To optimise the presentation of the measured functions and values, you may adjust the horizontal and vertical axes' graduations (X-RANGE and Y-RANGE, respectively); the vertical axis top scale value (Y-MAX) and the horizontal axis minimum value (X-MIN).

Z-axis Cursor

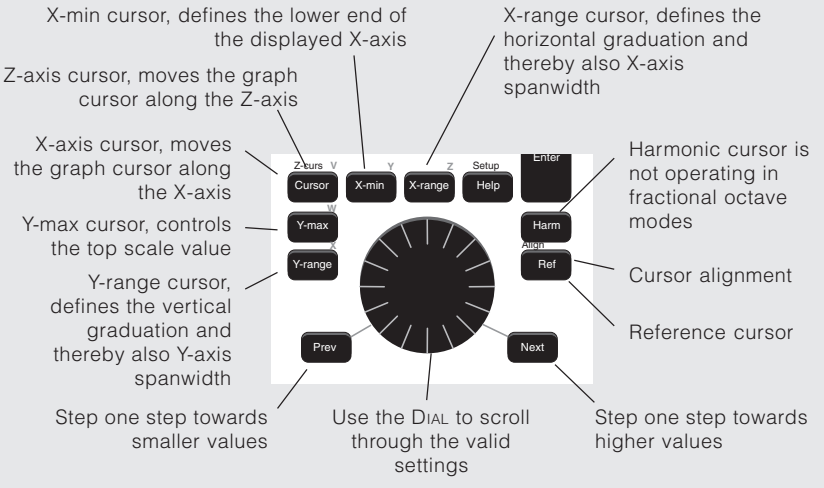
In multispectrum mode, the measured data can be represented by a three-dimensional matrix having level, frequency and time as the three dimensions.

If the display is set to display a spectrum, the X-axis will be the frequency axis, the Y-axis will be the level axis and the Z-axis will be the time axis.

You may also display the time profile for a certain frequency band (e.g. the 3150Hz 1/3 octave band or the A-weighted value). In this case the X-axis becomes the time axis and the Y-axis remains the level axis, while the Z-axis now becomes the frequency axis.

To switch between time and frequency as X-axis, use the LF/LT key.

The Cursor Control Keys


X-min cursor, defines the lower end of the displayed X-axis
X-range cursor, defines the horizontal graduation and thereby also X-axis spanwidth
Z-axis cursor, moves the graph cursor along the Z-axis
X-axis cursor, moves the graph cursor along the X-axis
Y-max cursor, controls the top scale value
Y-range cursor, defines the vertical graduation and thereby also Y-axis spanwidth
Harmonic cursor is not operating in fractional octave modes
Cursor alignment
Reference cursor
Step one step towards smaller values
Use the DIAL to scroll through the valid settings
Step one step towards higher values
Z-curs
Cursor
X-min
X-range
Setup
Help
Enter
Y-max
Y-range
Harm
Align
Ref
Prev
Next

Note that the Y-axis is left unaffected by this swopping, since the level in both cases will be the same (RMS or Peak values, always in dB).

To move along the X-axis (irrespective of whether this represents time or frequency) press CURSOR and then use the DIAL or the PREV & NEXT keys.

To move in the Z-direction (again irrespective of what it represents) press 2ND CURSOR and do as for cursor movements along the X-axis. A small icon in the display will indicate that Z-cursor is selected.

3D Cursor Functions

Measured data acquired in multispectrum mode may also be displayed as a three-dimensional graph. For sound intensity functions, only positive values can be shown.

To activate this feature make a multispectrum measurement, set the column 1 of the *Displayed curves* in the Display Setup menu to one of the functions actually measured and press the 2ND LF/LT keys.

If you fail to set up a function actually measured all you'll see will be an empty floor and no graph.

The graph is always drawn so that the periods most recently acquired appear to be closest to you.

Use the LF/LT to flip the X- and Z-axis.

Whilst the graph is being drawn the TYPE key is used to select the next display type. The CURSOR with or without 2ND, the X-MIN and the X-RANGE are used to select the cursor function, to which the DIAL and the PREV & NEXT keys apply. However, these step control keys will not cause response until the graph has been redrawn completely.

When the keys Y-MAX; Y-RANGE; 2ND X-RANGE and 2ND Y-RANGE are pressed prior to using the step control keys, any display update in progress will be aborted and a new update started.

Cursor Alignment

In dual display mode the graph cursors (X-axis cursors) of the two windows may be aligned with each other and moved together in either direction.

Press 2ND REF to align cursors. To deactivate press again, press TYPE or change to another measurement mode (e.g. single spectrum or intensity).

The master cursor will be the cursor of the active window (the one surrounded by the thick frame).

The two display windows must contain data acquired with the same filter bandwidth (e.g. 1/3 octave bands), but they need not display data in identical domains – i.e. one can show data with time as X-axis while the other shows frequency as X-axis. The contents of the two windows need not belong to the same measurement.

Reference Cursor

The reference cursor is used to investigate the difference between two points on a graph.

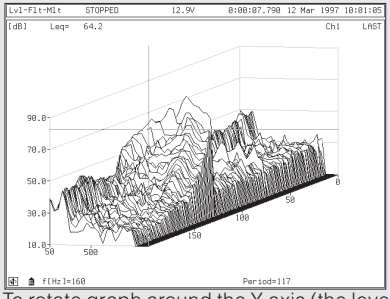
To activate the reference cursor, select CURSOR to move the graph cursor to your reference point, then press REF and use the DIAL or PREV & NEXT to move the graph cursor to a point elsewhere on the graph. The reference point will now be stated in a line inserted in the active window.

The Icons Show the Type of Cursor Selected

A B C D E F G H

A: X-min
B: X-range
C: Y-max
D: Y-range
E: Graph cursor
F: Z-cursor
G: Y-axis rotation (3D)
H: X-axis rotation (3D)

3D Cursor Functions



To rotate graph around the Y-axis (the level axis) press 2ND Y-RANGE and use PREV & NEXT or the DIAL. Origin of rotation is not origin, but a point exactly in the middle of the XZ- (time-frequency) plane.

To rotate the graph around the X-axis, press 2ND X-RANGE and use PREV & NEXT or the DIAL.

PREV & NEXT step size is 0.03 radians.

Remember to set the column 1 in the Display Setup menu to a function actually measured to get a graph. Only positive intensity values are shown.

Cursor Alignment

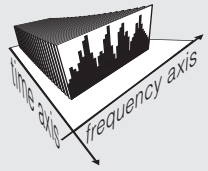
The master cursor is always the cursor of the active window (surrounded by the thick frame).

The two displays must contain data with identical filter bandwidth (e.g. 1/3 octave bands) but one can show spectrum while the other shows time profile.

The cursor will never move outside the master cursor's range, but if the period range of the master cursor window exceeds that of the other window, moving the graph cursor will cause the slave cursor to stop at its extreme end, while the master will go on until it reaches its extreme end. To spot when this occurs, watch the period number in each display window.

If the cursor alignment was active at the time you stored a setup file (a .cfg file) and you put this on a floppy disk naming the file 840.cfg, initialising the analyser with this floppy will cause the cursor alignment function to be set active as a part of the initialisation.

Tip: Do not confuse top scale and full scale! The top scale is purely a display control indicating the top of the vertical axis, while the full scale controls the input amplifier gain settings. Changing the latter will delete the Last register contents!



More 3D. If the X-axis represents the frequency, the Z-axis will represent the time and vice versa. Use LF/LT to swop.

Scaling and Graduation Range

Y-axis: 20, 40, 60, 80 or 100 dB across the vertical scale or in a 1–2–5 sequence when set to engineering units.

X-axis: 1:8, 1:4, 1:2, 1:1, 2:1, 4:1 and 8:1 applies to time displays only (not the fractional octave spectrum displays)

Data measured or retrieved can be shown either as graphs or tabulated. To switch between these two ways of presenting the data press the NUM. key.

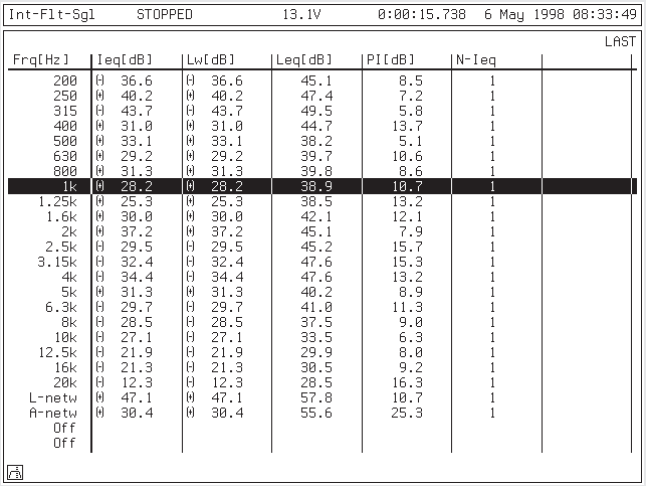
The contents of the table is determined by the Display Setup menu.

You navigate in the table by means of the *DIAL*, PREV & NEXT and the 2ND 9 (PGUP), 2ND 3 (PGDN), 2ND 7 (HOME) and 2ND 1 (END).

The table is no more than a numeric representation of the data, so the Z-cursor and cursor alignment works even here.

A Tabulated Example

Num.



Frq[Hz]	Ieq[dB]	Lw[dB]	Leq[dB]	PI[dB]	N-Ieq
200	A 36.6	A 36.6	45.1	8.5	1
250	A 40.2	A 40.2	47.4	7.2	1
315	A 43.7	A 43.7	49.5	5.8	1
400	A 31.0	A 31.0	44.7	13.7	1
500	A 33.1	A 33.1	38.2	5.1	1
630	A 29.2	A 29.2	39.7	10.6	1
800	A 31.3	A 31.3	39.8	8.6	1
1k	A 28.2	A 28.2	38.9	10.7	1
1.25k	A 25.3	A 25.3	38.5	13.2	1
1.6k	A 30.0	A 30.0	42.1	12.1	1
2k	A 37.2	A 37.2	45.1	7.9	1
2.5k	A 29.5	A 29.5	45.2	15.7	1
3.15k	A 32.4	A 32.4	47.6	15.3	1
4k	A 34.4	A 34.4	47.6	13.2	1
5k	A 31.3	A 31.3	40.2	8.9	1
6.3k	A 29.7	A 29.7	41.0	11.3	1
8k	A 28.5	A 28.5	37.5	9.0	1
10k	A 27.1	A 27.1	33.5	6.3	1
12.5k	A 21.9	A 21.9	29.9	8.0	1
16k	A 21.3	A 21.3	30.5	9.2	1
20k	A 12.3	A 12.3	28.5	16.3	1
L-netw	A 47.1	A 47.1	57.8	10.7	1
A-netw	A 30.4	A 30.4	55.6	25.3	1
Off					
Off					

Editing in the Tables

You may edit tabulated values except the Last register. Transfer data to another register before editing.

To start editing go to the line to edit and press 2ND AUX (EDIT). Selected position is then shown highlighted.

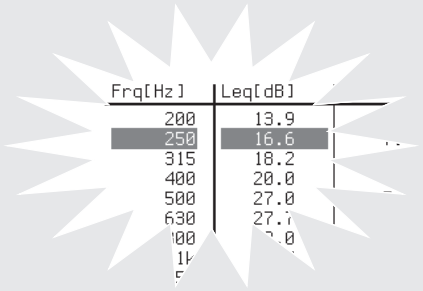
Use the numerical keypad to key in the new value. Terminate by ENTER or abort editing with ESC.

Use DIAL, PREV & NEXT and the 2ND 9 (PGUP), 2ND 3 (PGDN), 2ND 7 (HOME) and 2ND 1 (END) to move up and down in the table.

Use ❖ (Field cursor) – those pointing horizontally – to move between columns.

Press 2ND AUX (EDIT) to deactivate function.

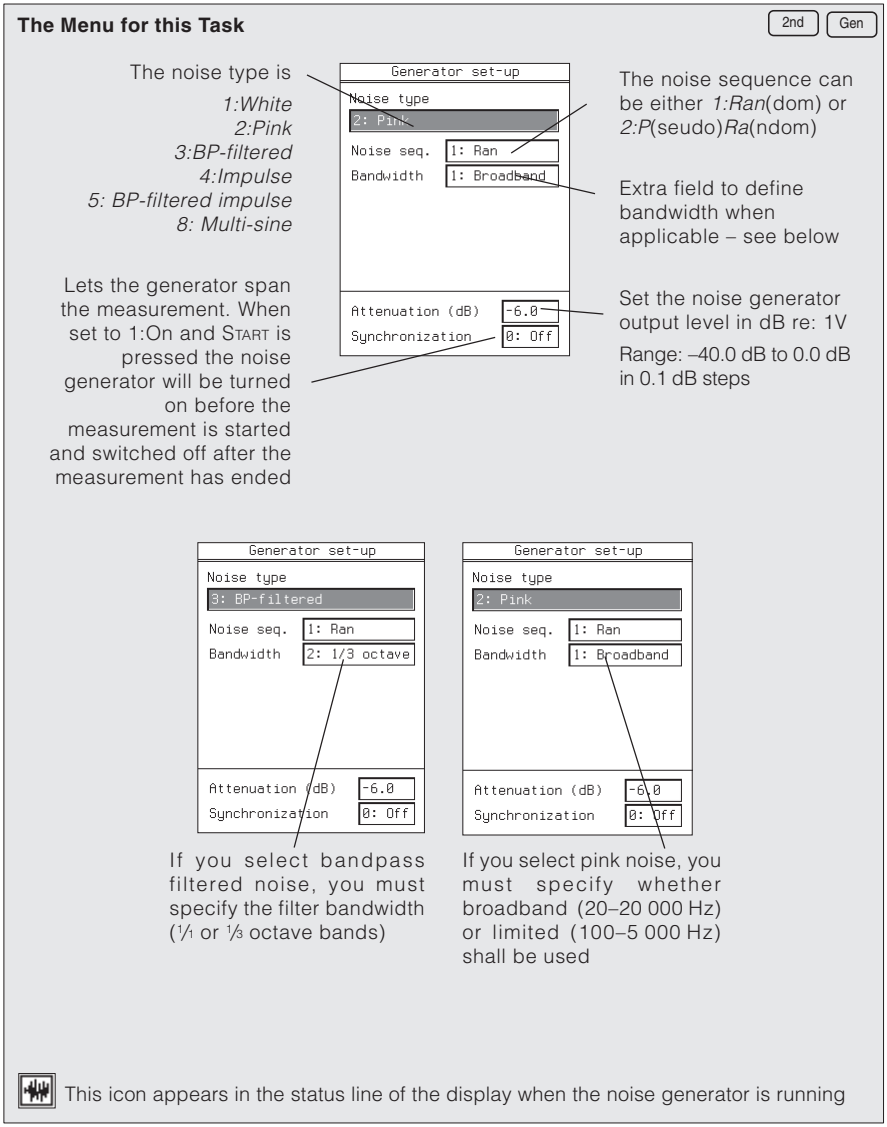
When You Edit in the Tables, it looks like this...



Frq[Hz]	Leq[dB]
200	13.9
250	16.6
315	18.2
400	20.0
500	27.0
630	27.7

The Noise Generator in Sound Intensity Mode

The Menu for this Task


2nd
Gen
The noise type is
1:White
2:Pink
3:BP-filtered
4:Impulse
5: BP-filtered impulse
8: Multi-sine
Generator set-up
Noise type
2: Pink
Noise seq.
1: Ran
Bandwidth
1: Broadband
Attenuation (dB)
-6.0
Synchronization
0: Off
The noise sequence can be either 1:Ran(dom) or 2:P(seudo)Ra(ndom)
Extra field to define bandwidth when applicable – see below
Set the noise generator output level in dB re: 1V
Range: –40.0 dB to 0.0 dB in 0.1 dB steps
Lets the generator span the measurement. When set to 1:On and START is pressed the noise generator will be turned on before the measurement is started and switched off after the measurement has ended
Generator set-up
Noise type
3: BP-filtered
Noise seq.
1: Ran
Bandwidth
2: 1/3 octave
Attenuation (dB)
-6.0
Synchronization
0: Off
If you select bandpass filtered noise, you must specify the filter bandwidth (1/1 or 1/3 octave bands)
Generator set-up
Noise type
2: Pink
Noise seq.
1: Ran
Bandwidth
1: Broadband
Attenuation (dB)
-6.0
Synchronization
0: Off
If you select pink noise, you must specify whether broadband (20–20 000 Hz) or limited (100–5 000 Hz) shall be used
This icon appears in the status line of the display when the noise generator is running

The Nor-840 comes with a powerful noise generator built in.

To access the setup menu press 2ND GEN.

The generator can be set up to supply white noise – whose level will increase by 3 dB per octave when viewed using fractional octave filters (but it looks flat when viewed using FFT because of its linear frequency axis); pink noise which looks flat in the fractional octave domain (but with a level decreasing by 3 dB per frequency doubling when viewed using FFT); bandpass filtered (1/1 or 1/3 octave bands only); impulse (1/1 or 1/3 octave bands only) and bandpass filtered impulse (1/1 or 1/3 octave bands only).

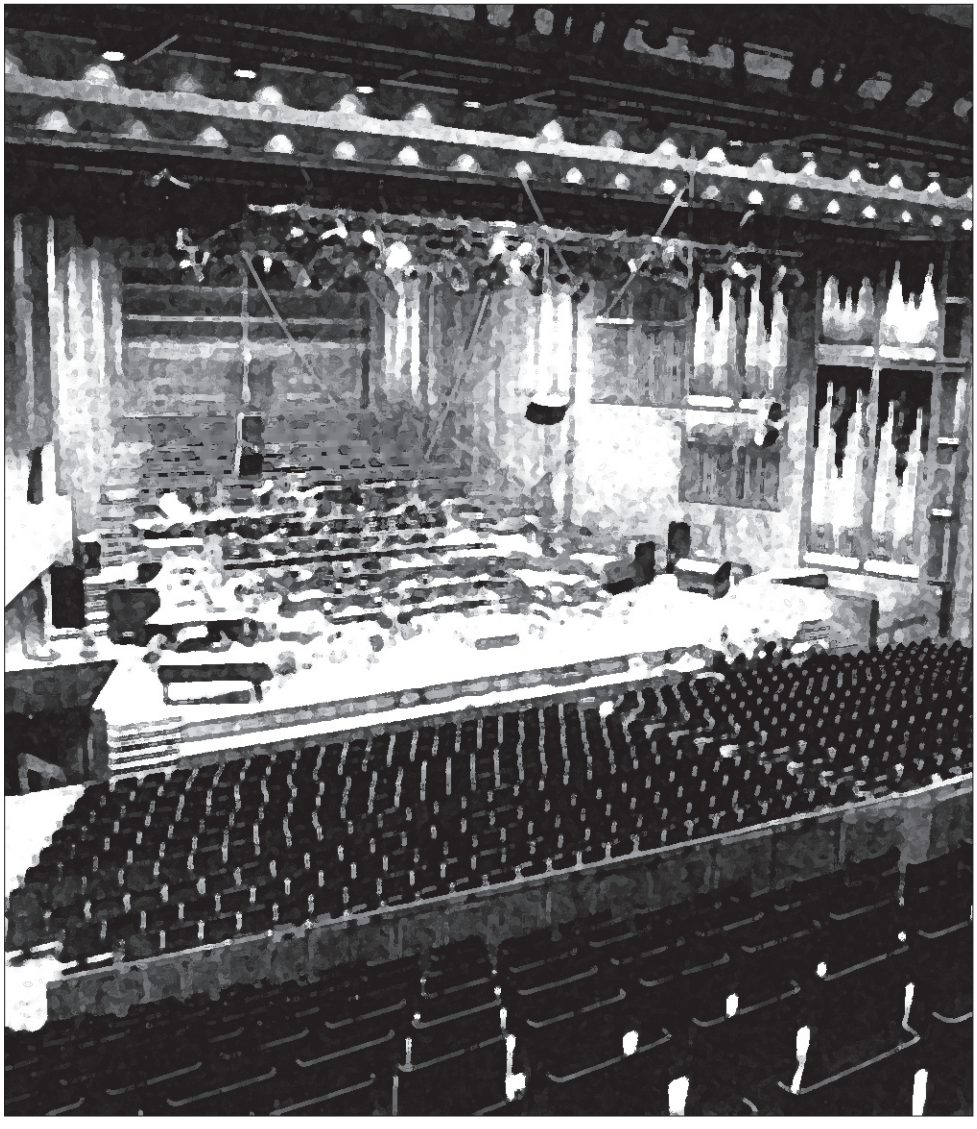
For the bandpass filtered signals, the frequency (band) is determined by the current graph cursor position.

Once the generator has been properly set up and the setup menu closed, press GEN to activate the generator and again to deactivate.

What Is Multi-sine?

Multi-sine is a complex signal consisting of sinusoids. The frequency components of this signal coincides with the lines of the FFT spectrum. Hence there is no signal energy outside these lines. Such a signal can be shown to have a periodicity equal to the time buffer length. This means that rectangular weighting can be applied without introducing discontinuities at the extremes.

In addition the phase of each of the sinusoids has been adjusted to give a crest factor much lower than that of conventional noise – approximately 1.3 (a single sinusoid has a crest factor of 1.4142, i.e. √2 or the same as 3 dB).



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92 MLS Measurement Principles and Features
94 Setting up for MLS Measurements
95 The Noise Generator in MLS Mode

Maximum Length Sequence

Maximum Length Sequence Fundamentals

The technique of *maximum length sequence* (MLS) is based on a few well-known facts. These are:

- The measurement object is excited by means of impulses
- Any deformation (time-smearing) of the impulses recorded at the output side of the measurement object originates from what the measurement object did to these impulses
- Repeated excitation is used to average out background noise

Applies to System Analysis

The MLS technique is used for *system analysis* and not *signal analysis*. Signal analysis deals with signals while system analysis aims at revealing system specific properties. Properties which are independent of the signal used to excite the system.

As system analysis requires knowledge about both the stimulus and the response of a system to be able to characterise it, is will always be a two- (or more) channel analysis. You cannot make system analysis with only one channel – measuring the stimulus only, will tell you nothing about the response and hence how the system behaves. Likewise, measuring the response only, will leave the stimulus unknown and again no system information will be revealed.

Nevertheless, MLS is applied to reverberation time measurements with great succes. How come it may be applied to such measurements? After all, reverberation time measurements are single channel measurements. According to the above they should belong to the signal analysis type of measurements rather than the system analysis type.

The clue here is that we know the stimulus – it is impulse or noise excitation – and we make use of this knowledge. Therefore we actually have two channels.

For impulse excitations we assume that the impulse is so close to a perfect impulse that we skip the measurement of it – but we still make use of it.

For noise excitations the only requirement is that we excite the measured frequency bands with a noise whose level is sufficiently high above the background noise level to generate decays that work. Hence we do not need to measure the stimulus in this case either (with the exception of the initial level to determine when to stop calculating).

The same thinking applies to MLS measurements also. Since we know the stimulus, we can concentrate on the response and MLS becomes a single channel analysis, albeit system analysis.

A Train of Impulses

The noise generator of the Nor-840 has been designed to output a very special signal when used with the MLS extension. The signal consists of a train of impulses, each of the same amplitude, normalised to 1, but with the polarity varying in a certain pattern – ie. the amplitude is then +1 or –1).

The spectrum of the MLS signal looks like white noise, but it doesn't have the statistical properties of a white noise signal, since the amplitude is either +1 or –1 and therefore not Gaussian distributed.

The autocorrelation of this train of impulses is an impulse at τ=0, ie. again a sort of white noise look-alike.

When making sound insulation measurements, the impulses are fed to a loudspeaker placed in the sending room. Of course, loudspeakers are far from ideal, any loudspeaker, no matter how"high end"it may be, will inevitably distort the impulses.

However, for most practical purposes, the room will distort the impulses to such an extent that what the loudspeaker (and the microphone for that sake) does is without significance.

The Impulse Response

The distorted impulse is called the *impulse response* of the room since it expresses the way the room responds when excited with an impulse. When we excite the room with a train of impulses, we will end up with a train of impulse responses as well.

It can be shown that the impulse response the frequency response of a system is related through the *Fourier transform*. They form a pair in which both express exactly the same information, but in different domains; viz. time- and the frequency domain respectively.

The Nor-840 MLS excitation signal consists of period each containing 2^{17}–1 impulses equidistantly spaced along the time axis. The impulse frequency is a function of the highest frequency band employed in a measurement.

Time-shift and Summation

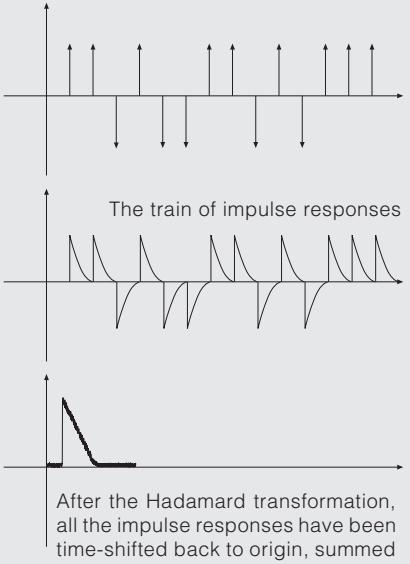
Now comes the clue; each impulse response is captured, retained by the analyser and then summed with the next impulse response to arrive. To make this work all the reponses are time-shifted (so that they seem to occur simultaneously and summed together).

A very convenient algorithm already exists for this purpose – known as the *Hadamard* transform.

Synchronous Averaging

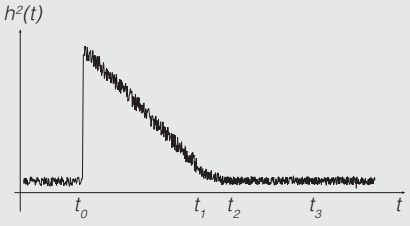
Normally, an MLS measurement consists of several periods each containing 2^{17}–1 impulses. All the periods are averaged together to form one period of 2^{17}–1 time-synchronously averaged impulse responses. The Hadamard transform will now time-shift all the 2^{17}–1 impulse responses back to the origin and calculate the averaged impulse response. This broadband (AC) impulse response is then fed to the filters of the Nor-840 to obtain an impulse response for each fractional octave band. *Note that the MLS extension applies to Level mode of the Nor-840 only.*

The excitation signal is a train of impulses with amplitude +1 or –1 (ie. they are equal in magnitude and normalised to a certain value, e.g. 0.775V ≡ 1)



After the Hadamard transformation, all the impulse responses have been time-shifted back to origin, summed and properly scaled to yield a single impulse response.

The squared impulse response obtained with MLS. The t_1 is where the noise starts to dominate, while t_2 and t_3 are used to determine the noise energy…



Single vs. Multispectrum

In single spectrum mode we calculate the mean signal energy to get the Leq, while in multispectrum mode we calculate short-time Leq values (as usual) from the impulse response as if it were a regular impulse measured by the analyser in conventional multispectrum mode.

Therefore, the MLS extension acts as a shell on the top of the rest of the analyser. The rest of the analyser is happily ignorant about how the signal measured was generated.

The Fig. above shows an impulse example – the impulse has been squared and converted to the dB domain.

A *noise density* can be determined from the background noise measured between t_2 and t_3. Since the noise density is the same all along the time axis, we are now able to determine the noise contents in the signal for all values of *t*. *This means that we are able to determine the true signal-to–noise ratio for the impulse response itself!*

In addition, since practically all the signal energy is located to the left of t_1 in the Fig. and since the noise is distributed evenly along the time axis, this method exhibits a signficant improvement in the signal-to–noise ratio. Why? Simply because the signal remains concentrated while the noise is spread out.

This means that the MLS method makes it possible to make measurements with signal-to–noise ratios hitherto unusable.

The snag is that you trade requirements for dynamics for requirements for more time spent, but in extreme cases you will still appreciate it.

Would you like to read more?

A more detailed discussion of the MLS can be found in O.H. Bjor and B. Winsvold, "Deterministic Excitation Signals Reduces Statistical Spread and Extraneous Noise Contamination in Sound Transmission Measurements", Proceedings of Inter-Noise 94, pp. 1469–1474.

Note: The measured levels have already been corrected for the apparent background noise. This follows from the algorithm itself. No further corrections must be made!

The differences between a conventional measurement and an MLS measurement are few, but significant. This article will elaborate somewhat on the practical aspect of these differences.

The MLS mode works with Level mode only; it cannot be used with other modes.

Setting It All Up

Although the MLS can find its way through extraneous noise much better than conventional methods can, the method will still benefit from a high noise generator output level.

Hence, we advise you to set it as high as possible, but be sure to check that the loudspeaker/power amplifier combination works well within its linear region.

Unlinearities are best spotted by making an MLS multispectrum measurement and then investigating the noise part of the impulse response for spikes – cf. the examples in the Fig. on this page spread.

Apart from the parameter settings required for any Level mode measurement, you may also specify the signal-to–noise (S/N) ratio needed. As a general rule an S/N ratio of 6 dB should do fine for most level measurement purposes, while a minimum of 40 dB should be used with RT measurements. See also *Setting up for MLS measurements* for more on this.

Making MLS Measurements

MLS measurements are made in the same way as measurements are made in conventional Level mode. The MLS has been put as a shell on the top of Level mode, preparing the measured data so that Level mode can make use of them directly. In this way no reconfiguration of the Level mode is needed and no deep user-knowledge of the MLS process is needed either.

Just set up the analyser as you would do for a Level measurement. The only difference is – as already mentioned – that you may specify a required signal-to–noise ratio.

The measurement is started in the usual manner, the only extraordinary now is that the analyser runs through a complete MLS signal period to ensure that the actual measurement takes place after steady-state conditions have been established.

If you pause the instrument before the preset number of averages has been reached, you may resume the measurement as usual.

However, you may also enter the measurement setup menu during the pause and change the number of averages required .

Make no Corrections for Background Level

Note that the MLS method compensates for the background noise by itself. Do not introduce any further compensation, which you may be used to with conventional measurements.

Leaving the MLS Mode

To leave the MLS mode, press 2ND ANALYSE and select *Normal*.

Alternatively, switching the analyser off will also cause the instrument to leave the MLS mode.

Saving the Impulse Response on Disk

The broadband impulse waveform may be stored on disk. The file cannot be read back into the analyser, but any PC (the optional of the Nor-840 itself included) equipped HP's SDF utilities or similar software will be able to make use of it. Details to the right.

Display Features

The only extras provided by the MLS extension is ability to display the apparent S/N ratio with and without MLS. These are, of course, theoretical considerations serving to provide information on what you gain from using MLS compared to traditional methods. The S/N options appear in the numerical table setting in the display setup menu. Annotation used is S/N M for the S/N ratio with MLS and S/N N (for Normal) without MLS. For multispectrum measurements the feature is not available while for reverberation time measurements S/N M is the only ratio available.

Time Reversal

The MLS process generates an impulse which is fed to the analyser's Level mode and analysed there as if it were any measured impulse.

However, since the impulse is stored in the RAM of the analyser, it may be reversed in time ("played backwards") before it is presented to the Level mode.

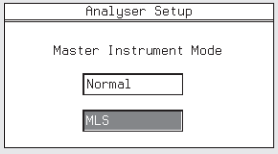
The advantage of doing this is that the analyser's minimum reverberation time

limits will be significantly reduced compared to the conventional methods.

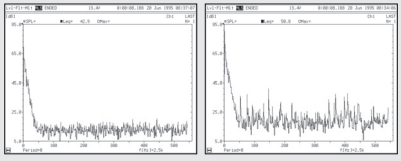
Time reversal is set up in the reverberation time setup menu, which appears slightly redesigned when MLS is active.

Entering and Quitting MLS Mode

Press 2ND ANALYSE to produce this dialogue box. Select mode



Spotting Unlinearities



Unlinearities tend to show up as spikes in the noise part of the impulse response. Shown here are two measurements, both 10 averages. The left was made with loudspeaker/amplifier working in their linear regions, the right while the loudspeaker was severely overloaded.

Saving the Broadband Impulse Response on Disk

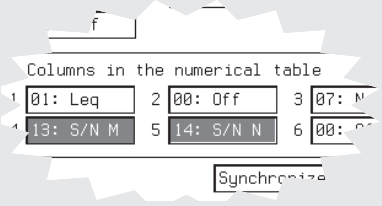
To save the impulse response (waveform) on disk do as when storing any file on disk, but set the *Type* to *4: Imp. Resp.*

Period Length of the MLS Signal Is Bandwidth-dependent!

The MLS sequence length is fixed to 2^{17}–1 samples. It is possible to change the period duration (needed for *inter alia* RT measurements) T_p by changing the sampling frequency f_s. When changing fs we must satisfy the Nyquist sampling theorem, the upper frequency band will therefore determine the sampling frequency as follows:

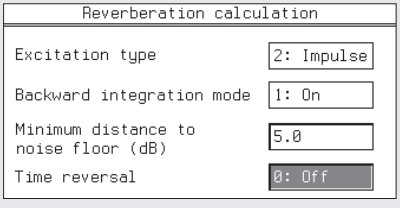
Uppermost frequency band(f_h)	Sampling frequency(f_s)	MLS sequence period length (T_p)
12 500–20 000 Hz	64 000 Hz	2.048 s
6 300–10 000 Hz	32 000 Hz	4.096 s
3 150–5 000 Hz	16 000 Hz	8.192 s
1 600–2 500 Hz	8 000 Hz	16.384 s
800–1 250 Hz	4 000 Hz	32.768 s
0.1–630 Hz	2 000 Hz	65.536 s

Displaying the Signal-to–noise Ratio



The only extras provided by the MLS extension is ability to display the apparent S/N ratio with and without MLS. The options appear in the numerical table setting in the display setup menu. Annotation used is S/N M for the S/N ratio with MLS and S/N N (for Normal) without MLS. For multispectrum measurements the feature is not available, while for reverberation time measurements S/N M is the only ratio available.

Time Reversal in Reverberation Mode


Reverberation calculation
Excitation type 2: Impulse
Backward integration mode 1: On
Minimum distance to noise floor (dB) 5.0
Time reversal 0: Off

For reverberation time calculations, the impulse response generated by the MLS process may be reversed in time to enable measurements of very short reverberation times. Set the shown parameter field to *On* to activate this feature and back *to Off* to deactivate it.

All registers are cleared!

Note that switching to MLS mode causes the analyser to reset it self completely. This means that the contents of all registers will be deleted upon entering MLS mode.

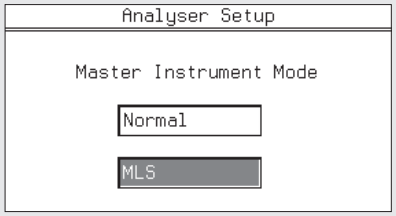
The MLS technique can be used with Level mode, single spectrum or multispectrum and with Reverberation time mode.

You enter MLS mode by pressing 2ND ANALYSE. Select *MLS*.

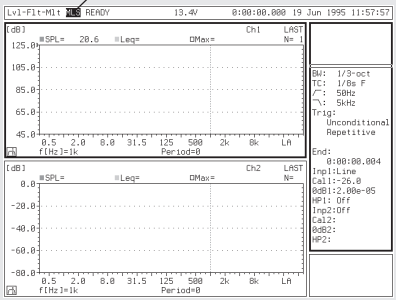
To return to normal (or conventional) mode press 2ND ANALYSE again and select *Normal*.

Alternatively, you may switch the analyser off. When you switch it on again it will enter conventional single spectrum Level mode. Any data not stored will then be lost, however.

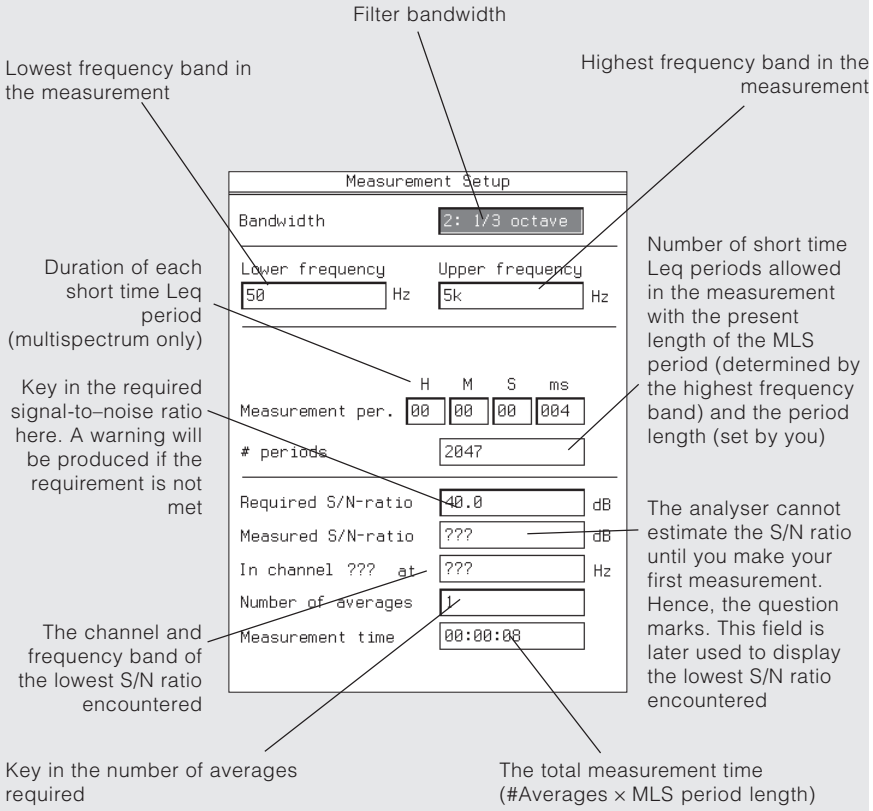
Entering MLS Mode



The status bar of the analyser shows the word MLS in reversed video to warn you:



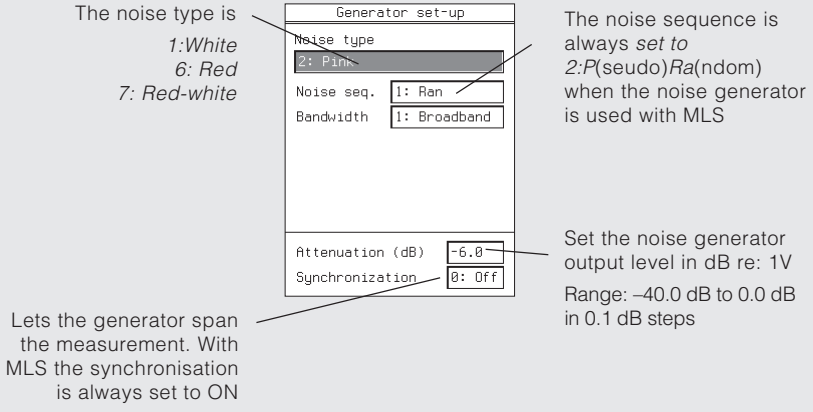
The Menu for this Task


Filter bandwidth
Lowest frequency band in the measurement
Highest frequency band in the measurement
Measurement Setup
Bandwidth
2: 1/3 octave
Lower frequency
Upper frequency
50
Hz
5k
Hz
H M S ms
Measurement per.
00 00 00 004
periods
2047
Required S/N-ratio
40.0
dB
Measured S/N-ratio
???
dB
In channel ??? at
???
Hz
Number of averages
1
Measurement time
00:00:08
Duration of each short time Leq period (multispectrum only)
Key in the required signal-to–noise ratio here. A warning will be produced if the requirement is not met
The channel and frequency band of the lowest S/N ratio encountered
Key in the number of averages required
Number of short time Leq periods allowed in the measurement with the present length of the MLS period (determined by the highest frequency band) and the period length (set by you)
The analyser cannot estimate the S/N ratio until you make your first measurement. Hence, the question marks. This field is later used to display the lowest S/N ratio encountered
The total measurement time (#Averages × MLS period length)

Note: When switching to MLS mode the contents of all the registers will be cleared!

The Menu for this Task

2nd Gen


The noise type is
1:White
6: Red
7: Red-white
Generator set-up
Noise type
2: Pink
Noise seq.
1: Ran
Bandwidth
1: Broadband
Attenuation (dB)
-6.0
Synchronization
0: Off
The noise sequence is always set to 2:P(seudo)Ra(ndom) when the noise generator is used with MLS
Set the noise generator output level in dB re: 1V
Range: –40.0 dB to 0.0 dB in 0.1 dB steps
Lets the generator span the measurement. With MLS the synchronisation is always set to ON

This icon appears in the status line of the display when the noise generator is running

Red-white Noise Cut-off Frequencies

The red-white noise is a combination of the two noise types to boost the spectrum extremes for a better overall signal-to–noise ratio. The signal is red in the lower end and white in the upper end of the spectrum, making it look like a "V" when viewed using a logarithmic frequency scale. The frequency at which the spectrum changes from red to white is the geometric mean of the frequency range of the measurement (which you set in the measurement setup menu, for example 50–10 000 Hz).

The geometric mean on a linear scale corresponds to the arithmetic mean on a logarithmic scale, so what we actually do is that we divide the frequency range in two halves and let the lower part (the left part in the Nor-840 display) use red noise and the upper part (the right part in the display) use white.

Your Nor-840 comes with a powerful noise generator built in.

To access the setup menu press 2ND GEN.

In MLS mode the generator can be set up to supply *white noise* – whose level will increase by 3 dB per octave when viewed using fractional octave filters; *red noise* which will decrease by 3 dB per octave when viewed in a similar manner or a combination of the two called *red-white noise*.

The main purpose of the red-white noise is to boost the extremes of the spectrum to obtain a better overall signal-to–noise ratio.

Once the generator has been properly set up and the setup menu exited, press GEN to activate the generator and again to deactivate.

98 Spectral Weighting Function Fundamentals
99 Creating Spectral Weighting Functions from Measurements
100 Creating Spectral Weighting Functions from Scratch
101 Loading a Spectral Weighting Function into the Analyser
102 Applying a Spectral Weighting Function to a Measurement

Generating Spectral Weighting Functions

The NOR-840 lets you define your own spectral weighting functions. These can be directly based on an actual measurement, or they can be keyed in by you.

This feature applies to fractional octave modes only.

Your analyser comes with the following set of predefined spectral weighting functions as standard:

- *SumA, SumB, SumC* and *SumL* are spectral weighting functions used to calculate the weighted value based on the frequency range actually measured. For frequency ranges smaller than the standards define for the A-, B- and C-curves, the value calculated will differ from what the value would be if measured with a separate A, B or C filter-network. For L(inear) the same will apply if significant amounts of energy exist in the signal outside the measured frequency range.
- *Hand-arm, Whole Body X-Y, Whole Body Z* and *Whole Body Combined.*
- *Sumf* is a sum of the energy of all the frequency bands measured with flat (or no) weighting.

There is a difference between linear and flat. Linear is defined as no spectral weighting between 20 Hz and 20 kHz and SumL is defined similarly. This means that for vibration measurements, the L and SumL networks are hardly usable at all. Therefore a flat spectral weighting network has been introduced, spanning the entire frequency range from 0.1 Hz to 20 kHz. This spectral network is available in synthesised form only and thus named *Sumf*.

Applications

Customised spectral weighting functions can be used as follows:

- As reference curves
- As a spectral weighting function applied to the displayed function (display pre-weighting)
- As a bargraph displayed to the right of the frequency spectrum representing the combined level of the measured frequency bands when the spectral weighting is applied. In this way more than one spectral weighting can be applied to the very same measurement.

Hence, the same spectral weighting function may appear in three places in the display at the same time, if you so wish.

Apply with Care!

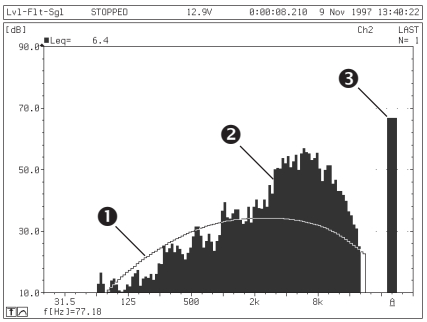
Synthesised spectral weighting functions will not always work as a true substitute for the real thing.

One example is the already mentioned A-, B- and C-weightings. Failure to apply the synthesised versions to a full range measurement will yield results inconsistent with what you would get using true separate filters of the same type.

Another example is the application of spectral weighting to vibration measurements. The accelerometers' output signal is acceleration. Sometimes velocity or displacement is required – either because standards or conventions require it, or simply because you may want a "flatter" spectrum, fitting better within the display range available. Integration of spectra is very simple in the fractional octave domain; an integration of a spectrum simply corresponds to multiplying the level (in *pascals*) of each frequency band by $1/2\pi f$, or by adding $20\log[1/2\pi f]$ to the level (in *dB*) to each frequency band. The frequency to be used should then be the centre frequency of each frequency band.

This works fine for stationary RMS signals. However, the procedure cannot be applied to peak signals or to transients, as both these phenomena involve the need for information on *when* things occurred (i.e. phase relationships) to be calculated correctly. Hence spectral weighting functions must always be applied with great care!

Also, it is of vital importance to observe the need for sufficient measuring time. To yield valid results, your measurement must have a duration long enough to provide the frequency resolution needed. The BT product must, of course, be at least 1, but the higher the better – a BT product of 5 or more is preferable.



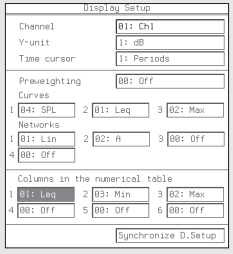
Customised spectral weighting functions may be applied in three ways; as reference curves ❶, as preweighting ❷ and as a bargraph ❸ representing the weighted level of the measured spectrum

Creating a spectral weighting function from a measurement…

❶ Select the register in which the measurement or the retrieved file resides

Example: Last or Aux register

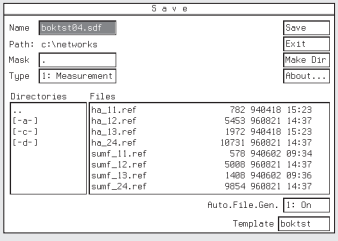
❷



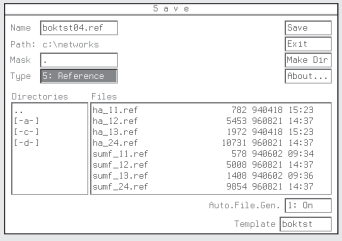
In the display setup menu select the function whose spectrum is to serve as spectral weighting function.

This is done by selecting this function as the one to appear in column 1 in the numerical table – see also the Tip below.

❸ Press COPY HDD



❹ Set *Type* to *Reference*



❺ Proceed as when storing measurements on disk

Tip for power users! Once your measurement has been made and the display setup menu has been set up with the function you want, just press COPY REF <Slot No.>, in which <Slot No.> is 1~8. You will now have generated a file whose file name is 840ref_x in which x is 1~8, located in the directory c:\840ref (root directory on the a: drive for units without hard disk). Each successive slot will overwrite an existing ref file.

Tip: The function whose spectrum is to serve as spectral weighting function is taken from column 1 in the numerical table setup (display setup menu). If column 1 is set to Off, the analyser will search the columns for activate functions. The first active found will then be used.

Tip: When you use the above method, the analyser generates a file similar to the one described overleaf – a synthesised reference file. With a text editor (or word processor) you may therefore open an existing reference file regardless of how it was made (from a measurement as here, or synthesised as described overleaf) and edit it *ad libitum*. Remember to save it as an ASCII file!

A spectral weighting function can be applied to a spectrum to show the effect of the weighting, to provide the weighted level or to serve as a reference spectrum to which the current measurement is compared.

The spectral weighting function is generated from a reference file.

The easiest way to generate a reference file is by using an actual measurement.

A reference file can be made from any type of measurement, but the direction information is not retained when reference files are generated from intensity measurements.

For intensity measurements the spectral weighting function will be shown symmetrically about the frequency axis.

The bandwidth of a spectral weighting function must be the same as the bandwidth of the spectrum to which it shall be applied. Apart from that no restriction applies – a spectral weighting function generated in multispectrum intensity mode may just as well be used in level single spectrum mode.

In multispectrum mode, the actual spectral weighting function will be the function set in column No. 1 of the numerical table for the period No. currently displayed in the active window. Hence regardless of whether the spectral weighting function was generated in single- or multispectrum mode, the result will be a single spectral weighting function – *always!*

This article describes how to create spectral weighting functions directly from an actual measurement. How to apply a spectral weighting function to a measurement is described on the following pages.

Creating Spectral Weighting Functions from Scratch

A spectral weighting function can be applied to a spectrum to show the effect of the weighting, to provide the weighted level or to serve as a reference spectrum to which the current measurement is compared.

The spectral weighting function is generated from a reference file.

The reference file is made either directly from a measurement as described on the previous page or from scratch, i.e. by keying in the value for each frequency band.

However, you may start by generating a reference file from a measurement and later modify this file manually.

To synthesise a reference spectrum entirely from scratch you will need a text editor such as NotePad (which is included with Windows™ or a wordprocessor program such as Microsoft® Word or WordPerfect™ from Corel®.

If your NOR-840 is equipped with the MS-DOS-extension, any text editor installed here may be used.

Follow the guidelines given here and save the file as ASCII text. The file must be given the extension *.ref*. Observe that a reference file is made for a certain bandwidth only and cannot be used with other bandwidths than the one it was designed for.

There is no need for any kind of END statement in the file.

An example of a synthesised reference file...

```
A  [840]
B  ;This file describes a weighting network for the NOR-840
   ;comments are preceded by ';'
   ;line length must not exceed 80 characters
C  Type = Reference
   Bandwidth = 1/3; may also be 1/1 or 1/12 or 1/24
   Dataformat = dB; may also be Lin or Sec (for RT)
D  [Data]
   -36.00; 0.10Hz
   -31.99; 0.125Hz
   -27.99; 0.16Hz
   -23.99; 0.20Hz
   -20.01; 0.25Hz
E  -16.05; 0.315Hz
   -12.18; 0.40Hz
   -8.51; 0.50Hz
   -5.27; 0.63Hz
     .      .
     .      .        (Some lines have been removed for clarity)
     .      .
   -160.03; 12.5kHz
   -166.03; 16kHz
   -172.03; 20kHz
```

A The file must start with this line typed exactly as shown

B Comments may appear everywhere, provided they are preceded by a semicolon

C Must always be present and located here (i.e. after the line `[840]` and before the line `[Data]`. However, the internal order of appearance of these items is not important. `Type` is always `Reference`, `Bandwidth` is `1/1`, `1/3`, `1/12` or `1/24` octave band, `Dataformat` is `dB`, `Lin`(ear values or `Sec`(onds) for reverberation time measurements

D This line, typed exactly as shown, must appear exactly where shown

E There must be a value (coefficient) for each frequency band all the way from 0.1 Hz to 20 kHz (both extremes included), e.g. 18 for 1/1 octaves and 54 for 1/3 octave bands etc.

Valid Ranges for the Coefficients...

Dataformat = dB: –327 to +327

Dataformat = Lin: 0 to 2.4e16 (2.4×10^{16})

Dataformat = Sec: 0 to 327

Note: When you use Dataformat = dB, neither attenuation nor amplification entered as 0 (zero). Amplification is entered as a positive dB value and attenuation as a negative.

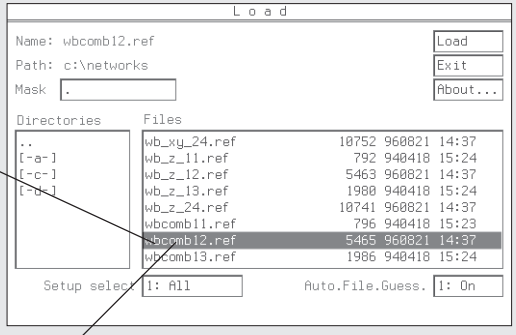
When you use Dataformat = Lin, neither attenuation nor amplification entered as 1 (one). Amplification is entered as a gain factor (>1 and positive) and attenuation as an attenuation factor (<1 and positive).

The tools for this task…

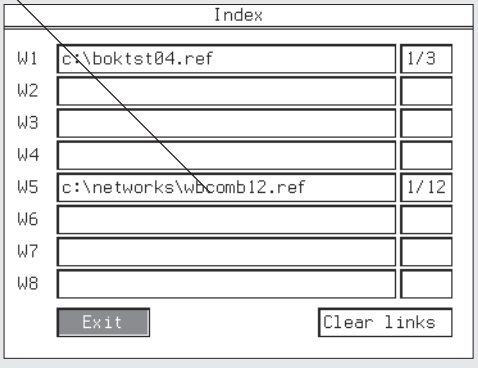
❶ Press HDD to produce the load menu

❷ Locate the file to be loaded

❸ Press 1–8 on the numerical keypad to load the file as W1–W8 and press ENTER to execute



Example: By pressing 5 followed by ENTER in the above menu, the selected file will appear as W5



This list of which files are used as which spectral weighting function number is available by pressing the INDEX key

Once generated, a reference file may be applied as spectral weighting function to any fractional octave measurement using the same filter bandwidth as the file itself does.

The first step – when you are going to apply a spectral weighting function to a measurement – is to load the corresponding reference file into the analyser.

Altogether, the analyser can hold 14 spectral weighting functions simultaneously, out of which 6 are predefined and the remaining 8 are entirely at your disposal.

These 14 spectral weighting functions are: Off (none); Lin; A; SumA; SumB; SumC; SumL and W1–W8.

You may have as many spectral weighting functions as you like on the hard disk or floppy disk, but only eight of them may be used simultaneously. These are denoted W1–W8.

Note: The analyser uses links instead of importing the reference file. This saves memory. However, every time you exit to MS-DOS mode and come back to analyser mode, the analyser will need to read the linked files. This will take time, particularly when the files are stored on floppy disk. You may not want this whenever you are not using the reference files. This is when to use the Clear links feature of the Index list.

Applying a Spectral Weighting Function to a Measurement

Applying a spectral weighting function (SWF) to a measurement can be done in three ways in the NOR-840.

An SWF may be used as:

- Spectral weighting function showing the measured spectrum weighted by the spectral weighting function – in the NOR-840 referred to as *preweighting*
- Bargraph indicating the total level of the spectrally weighted spectrum
- As a reference curve indicating how the measured spectrum looks compared to a certain reference.

The three options are independent of each other – the very same spectral weighting function may appear in all three positions at the same time!

The application of spectral weighing functions is a display matter only. It does not affect the measurement as such in any way.

There are two types of spectral weighting functions – preprogrammed and customised.

The preprogrammed functions are

- *SumA, SumB, SumC* and *SumL* which are spectral weighting functions that can be used to calculate the weighted value based on the frequency range actually measured. For frequency ranges smaller than the standards define for the A-, B- and C-curves, the value calculated will differ from what the value would be if measured with a separate A, B or C filter-network. For L(inear) the same will apply if significant amounts of energy exist outside the measured frequency range.

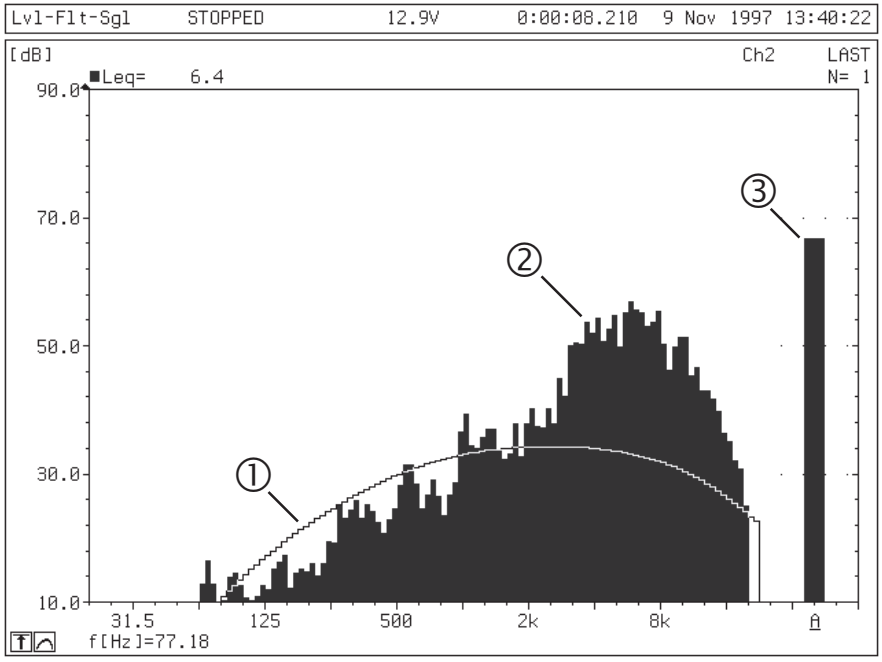
The customised functions are of three types.

- *Hand-arm, Whole Body X-Y, Whole Body Z* and *Whole Body Combined.* Comes as standard with the analyser, located in the directory c:\networks
- *Sumf* which is a sum of the energy of all the frequency bands measured with flat (or no) weighting. Comes as standard with the analyser, located in the directory c:\networks
- Your own reference files.

Note the difference between linear and flat. Linear is defined as no spectral weighting at all, but only between 20 Hz and 20 kHz and SumL is defined similarly. This means that for vibration measurements, the L and SumL networks are hardly usable at all. Therefore a flat spectral weighting network has been introduced, spanning the entire frequency range from 0.1 Hz to 20 kHz. This spectral network is available in synthesised form only and thus named *Sumf*.

Before you can utilise a spectral weighting function, the corresponding reference file must be loaded into the analyser. This is described on the previous page.

Customised spectral weighting functions may be applied in three ways; as reference curves ①, as preweighting ② and as a bargraph ③ representing the weighted level of the measured spectrum...



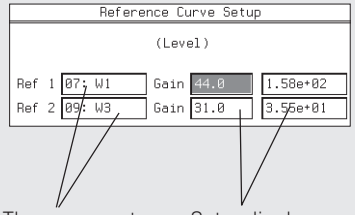
①

To get the spectral weighting function shown as a reference spectrum...

❶ Load the reference files you need as described on the previous page spread, if applicable.

Skip this point if you are going to utilise preprogrammed spectral weighting functions only.

❷ Press 2ND D.SETUP to produce the Reference Curve Setup menu.



There are up to two ref. curves available for each display window. Select the spectral weighting functions to be used as reference curves. (Example shows W1 and W3)

Set a display offset, i.e. gain for each curve. Set it in dB or absolute units as you like.

Changing one will change the other accordingly.

❸ Exit the menu

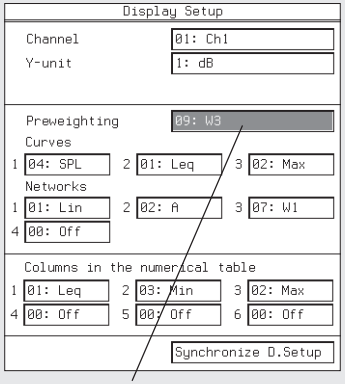
②

To get the measured spectrum shown weighted with the spectral weighting function (preweighting)...

❶ Load the reference files you need as described on the previous page spread, if applicable.

Skip this point if you are going to utilise preprogrammed spectral weighting functions only.

❷ Press D.SETUP to produce the Display Setup menu.



Set Preweighting to the spectral weighting function required (Example shows W3).

❸ Exit the menu

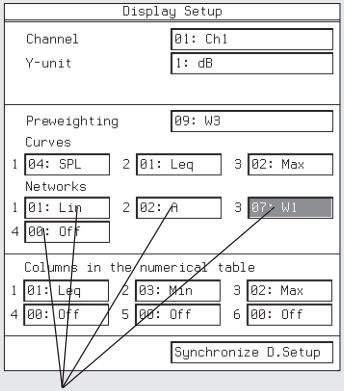
③

To get the level of the weighted spectrum indicated as a bargraph...

❶ Load the reference files you need as described on the previous page spread, if applicable.

Skip this point if you are going to utilise preprogrammed spectral weighting functions only.

❷ Press D.SETUP to produce the Display Setup menu.

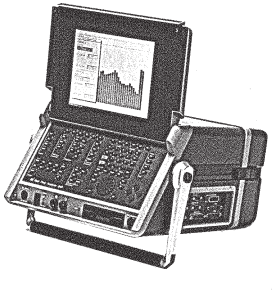


Set any number of the networks to the spectral weighting functions required – up to four simultaneously

❸ Exit the menu

Chapter 8

106 Memory Handling Fundamentals
108 Storing a Measurement on Disk
109 Autonumbering Files Stored Consecutively
110 Disk Handling Tools
111 Retrieving Stored Measurements
112 Storing Instrument Setups
113 Retrieving Instrument Setups



Memory Handling

The Nor-840 can store measured data as well as measurement setups.

A fully equipped Nor-840 has four registers and two external (although built-in) storage media. For simplicity we refer to all these as memory locations.

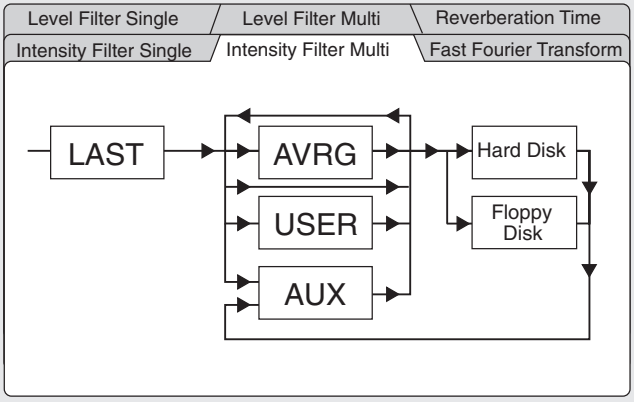
The registers are called Last; Average; Aux and User, while the storage media are the floppy disk and the optional hard disk.

There is one set of registers for each measurement mode of the analyser, as shown to the right. The registers are *not* nonvolatile, i.e. their contents are *not* retained if the analyser is switched off.

The following restrictions apply:

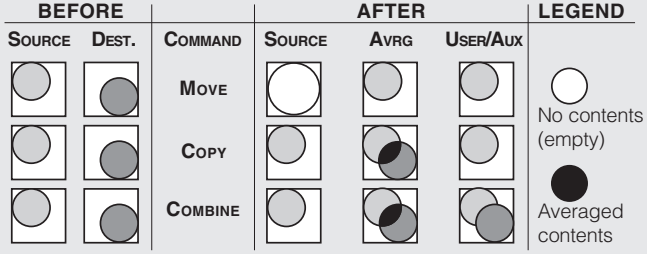
- The Last register is reserved for measurements only. Data can be copied *from* but not *to* this register, i.e. data cannot be transferred to the Last register – they have to come directly from a measurement.
- Data retrieved from any of the storage media enters the Aux register. From there they can be copied everywhere, except to the Last register.
- Averaging is done in the Average register only.
- All registers share a common register pool. A large amount of data stored in one mode may use so much register capacity that the storage capacity in other modes are seriously affected. Otherwise they are completely independent.
- Register contents are preserved when switching to another measurement mode. However, switching to MLS mode will clear all registers.

There is one register per instrument mode



Moving Data Between Memory Locations

The general command sequence is <SOURCE><COMMAND><DESTINATION>. SOURCE is where the data reside; COMMAND is MOVE, COPY or COMBINE while DESTINATION is the location to which the data will be moved (or copied).



The commands Move and Combine apply to the registers only, they do not apply to any of the external storage media (i.e. the disks).

Tip: If the contents of one register makes storing in other registers difficult (not enough free memory) clear some of the registers to make space. Remember to go to the corresponding measurement mode first. Save the contents on disk, if needed, before clearing.

Swopping Measurement Channel Contents

Press 2ND followed by LAST to swop the contents of channel 1 and channel 2.

Note that the swop takes place in the Last register only.

Clearing a Specific Measurement Channel – Leaving the Other Intact

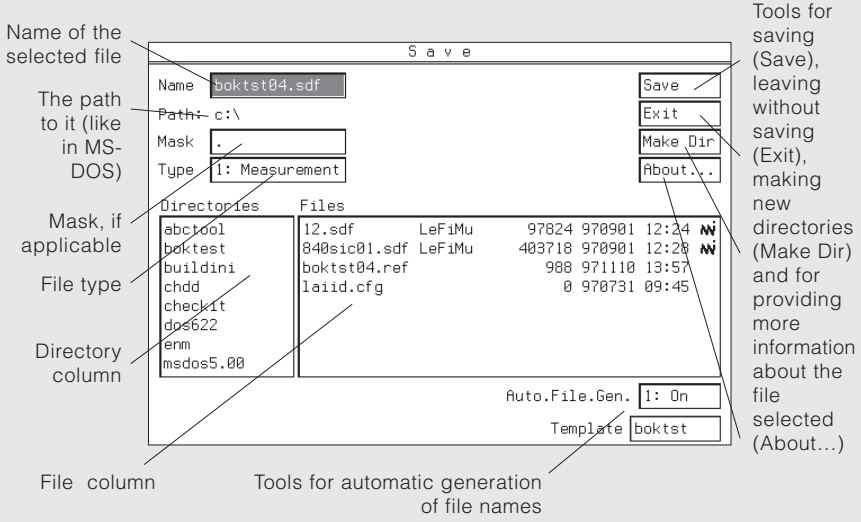
To clear the contents of a specific measurement channel of any register (Last register included) press

- CLEAR 1 <REGISTER NAME> to clear the contents of channel 1 of that register
- CLEAR 2 <REGISTER NAME> to clear the contents of channel 2 of that register

Clearing a Register

1. Go to the mode whose register is to be cleared (for example the intensity multispectrum mode)
2. Press CLEAR followed by the name of the register to be cleared (for example the Aux register).

Navigating in the Memory Handling Menus



Use the FIELD SELECTOR keys to go between the parameter fields. However, inside the directory or file column use PREV/NEXT or the DIAL. Although this is in strict accordance with the operating principles of the Nor-840, it may seem strange to some at first glance.

Masking out File Names

If your hard disk contains many different kinds of files, it may be useful to mask out other files than those applying to your present task.

To do this use wildcards. A wildcard acts a substitute for a name, a part of a name or an extension.

The Wildcard	Is used to	Example
Dot (.)	Display all Nor-840 measurement and setup files, but no other files	
Asterisk (*)	Represent a whole word or a group of characters, including no characters at all	*.pcx will display all files with the extension .pcx such as ole.pcx, arne.pcx, However, ar*e.pcx will display files like are.pcx, arne.pcx, arveee.pcx...
Question mark (?)	Represent a single character	ar?e.pcx will display files like arne.pcx, arve.pcx, but not files like avre.pcx, arpve.pcx...

Storing a Measurement on Disk

A fully equipped analyser will contain a built-in floppy disk drive (comes as standard on all models) and a built-in hard disk drive (must be ordered separately).

The storing procedure treats hard disk and floppy disk as variations on the same theme. The advantage of this is that the hard disk and the floppy disk then both are treated as external storage media. If, for example, your analyser is hooked up to a LAN (Local Area Network such as a Novell or NT Server network together with other computers) you will normally have access to several hard disks in addition to those in your analyser. All these disks will then appear as disks available for storing and retrieving alongside with the built-in ones and they are all treated in the same way.

If the measurement has been given a title, the eight first characters, ignoring any spaces, will be suggested as file name.

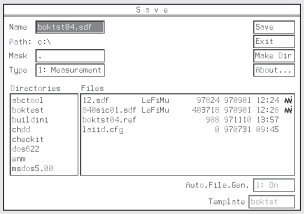
The File Extension .sdf

Regardless of what extension name you append to a measurement file, it will automatically be replaced by .sdf. This means that even if you specify no extension at all, the .sdf extension will still be appended by the analyser. This is done to make measurement files easier to locate upon file retrieval.

Saving a Measurement on Any Disk

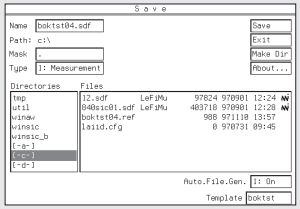
❶ Select the register where the data currently reside (for example the Average register, press AVERAGE to select this register)

❷



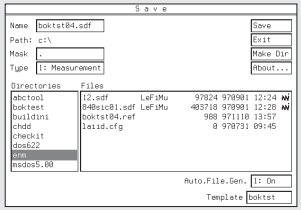
Press COPY HDD

❸



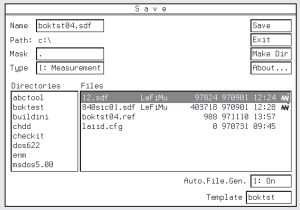
Select the required disk drive, press ENTER

❹



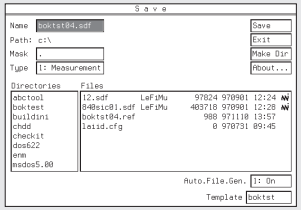
Position the cursor on the required directory and press ENTER to produce a list of the contents of the selected directory

❺



To overwrite an existing file, select that file to make the name appear in the *Name* parameter field and press ENTER

...or...

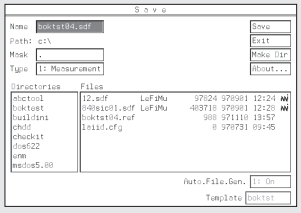


To save it under a new name, move up to *Name* and key in the file name

❻ Press ENTER twice or move to *Save* and then press ENTER to save the file

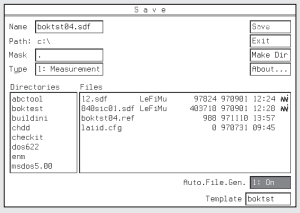
Setting up for Autonumbering

❶



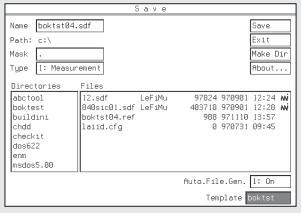
Press COPY HDD

❷



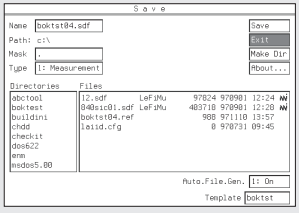
Set *Auto File Gen.* to *On*

❸



In the *Template* field, key in the common part of the file name – up to six alphanumeric characters. Key in the starting number of your number series at the end of the common name

❹



Select *Exit* and press ENTER to leave the menu, or select *Save* if you want to start using the feature immediately

Examples of File Autonumbering

- *hydro00*. Your number series will start from *00*. The first file will be named hydro*00*.
- *hydro*. This number series will also start from *00*, as no starting number was specified.
- *hydro2*. Again your number series will start from *00*, as no starting number was specified (starting number must be preceded by six alphanumeric characters to be recognised as a starting number. However, the first file will appear as *hydro200*, which is the file number *00* of *hydro2*.
- *hydro50*. Now your number series will appear as if it starts from *50*. The first file will be named *hydro50*, as it will be file number *0* of *hydro5*.
- *hydro134*. This time your number series will start from *34*, but it will appear as if it starts from *134*. *hydro1* is here the common part, the first file will therefore be *hydro134*.

When you do spatial averaging and want to retain the individual results or you make several (related) measurements, for example on large machinery, you may want to save the measurements using file names with a common part for easy identification later.

The Nor-840 offers the feature of automated generation of file names differing from each other by their last two digits only.

Examples of such file names would be *hydro00, hydro01, hydro02, …, hydro99*.

To make use of the feature, just store the measurement as usual and accept the file name suggested by the analyser. The next time a measurement is stored the last two digits will have been incremented. If this next file already exists, it will still be suggested, but if you have set the Help level to 2 or 3, you will be prompted to confirm whether the existing file is to be overwritten or not.

If the measurement has been given a title and the *Auto. File Gen* is set to *On* (was *Off*), the six first characters, ignoring any spaces, will be suggested as a file name template. In the Template field, the eight first characters will appear, but the last two will be replaced by the number series in the file name.

Getting File Information

Imagine that you are searching for a particular file and do not know exactly which file, or that you would like to increase the amount of free disk space available on the hard disk or a floppy disk. In order to avoid removing valuable files, you'll often need additional information about the files.

Extended file information is available via the *About...* parameter-field. This parameter-field is accessible from both the Save and the Load menu.

```
File Information

Measurement mode: Lvl-Flt-Mlt
Master Instrument Mode: Normal
No measurement title specified

Bandwidth       : 1/24-oct     Time constant   : 1/8s F
Lower frequency : 48.00Hz      Upper frequency : 4800.10Hz

Measurement start: Tue 22 Apr 1997 15:40:10
Measurement duration: 00:00:01:440
Period length: 00:00:00:020
Total number of periods: 72
Trigger condition: Manual

Channel 1                      Channel 2
Input           : Line         Input           : Line
Calibration     : -26.0dB      Calibration     : -26.0dB
Ref. Level (0dB): 2.00e-05     Ref. Level (0dB): 2.00e-05
Fullscale       : 95dB         Fullscale       : 100dB
High pass filter: Off          High pass filter: Off
```

The About... field of the Save and Load menus provide additional file information

What Is SDF?

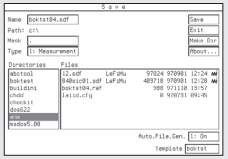
All measurement files are stored as SDF files in the analyser. The abbreviation SDF stands for Standard Data Format, a format designed by Hewlett-Packard (HP). The advantage of using the SDF format is that there are quite a few SDF utility programs available and these may be used to extract, postprocess and otherwise manipulate data stored in the Nor-840.

For details on Hewlett-Packard SDF utility programs, contact your local HP dealer.

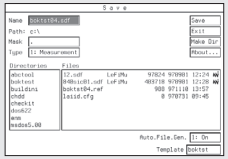
Creating a Directory

❶

Press COPY HDD

❷ 

Go to the directory in which you want to create a subdirectory

❸ 

Move up to *Name* and key in the name of the new directory

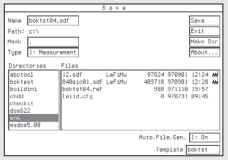
❹

Move to *Make Dir* and press ENTER

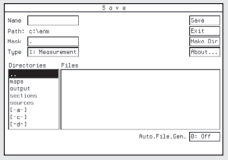
Deleting a Directory

❶

Press COPY HDD

❷ 

Go to the directory to be removed

❸ 

Make sure that the directory is empty (i.e. containing no files)

❹ Press the DEL key to remove the directory

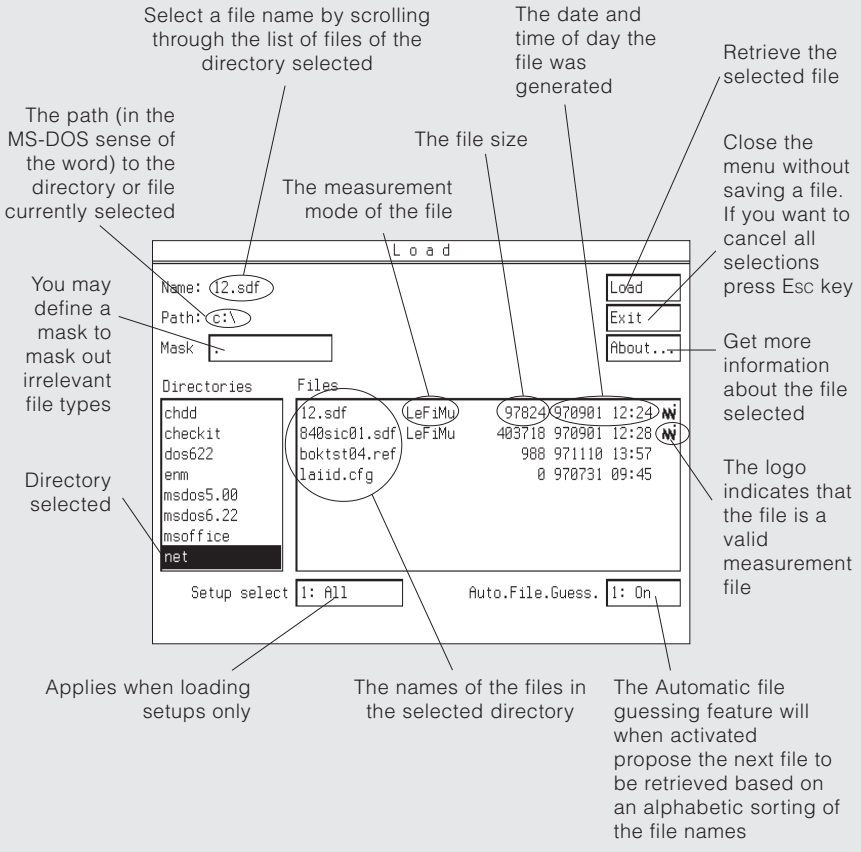
Deleting a File

❶ Press COPY HDD to produce the *Save* menu or just HDD to produce the *Load* menu

❷ Go to the file to be deleted

❸ Press DEL

The Menu for This Task


Select a file name by scrolling through the list of files of the directory selected
The date and time of day the file was generated
Retrieve the selected file
The path (in the MS-DOS sense of the word) to the directory or file currently selected
The file size
Close the menu without saving a file. If you want to cancel all selections press Esc key
The measurement mode of the file
Load
Name: 12.sdf
Path: c:\
Mask *.*
Load
Exit
About...
You may define a mask to mask out irrelevant file types
Get more information about the file selected
Directories
Files
chdd
checkit
dos622
enm
msdos5.00
msdos6.22
msoffice
net
12.sdf LeFiMu 97824 970901 12:24
840sic01.sdf LeFiMu 403718 970901 12:28
boktst04.ref 988 971110 13:57
laiid.cfg 0 970731 09:45
Directory selected
The logo indicates that the file is a valid measurement file
Setup select 1: All
Auto.File.Guess. 1: On
Applies when loading setups only
The names of the files in the selected directory
The Automatic file guessing feature will when activated propose the next file to be retrieved based on an alphabetic sorting of the file names

Retrieving Stored Data

❶ Press HDD to enter the Load menu

❷ Select the file to be retrieved. Use the *About...* function, if needed

❸ Select Load and press Enter or just press Enter from the *Files* scroll list.

When you retrieve stored measurements from the disk, they are automatically transferred to the Aux register of the mode in which the measurement was originally made. The analyser will switch to this mode and display the contents of this mode's Aux register. Observe that the display settings of the analyser will be applied to the retrieved file. This will include such things as the *displayed functions* which may cause your display to appear without graph if the display was set to display functions not contained in the retrieved file or if there is a severe mismatch between the retrieved levels and the current top scale setting. For FFT a beamfinder feature is available – see the FFT section of this manual for details.

Automatic File Guessing

If you are going to retrieve several files, one after the other, you may set the analyser to suggest the next file to be retrieved. This is achieved by setting the *Auto.File.Guess* to *On* in the Load menu. The file proposed will be the next file in the alphabet every time you press the Disk or the HDD key.

An obvious application for this will be when retrieving files with names generated by means of the autonumbering feature (described in the article *Autonumbering Files Stored Consecutively* on the previous page spread), but it may just as well be applied to any set of files.

The alphabetic order is such that the file AA01 will appear before AA02, the file A1A before A1B and the file 001 before 00A etc.

Storing Instrument Setups

You may want to store your favourite setups to save work the next time you use your analyser.

Setups are used to configure the analyser. A setup file is therefore a configuration file using the extension *.cfg* as the obvious identifier. This extension will be appended by the analyser.

When storing you must indicate whether the configuration file shall comprise all the measurement modes of the instrument or just the mode currently active.

If you store a setup file on a floppy disk under the name *840.cfg* and insert this floppy disk in the disk drive before turning on the analyser, the analyser will pick up this setup file and use it to configure the instrument as specified in the setup file.

If you are going to use the automatic file generation, you must store your setup file in the directory where you want to have your autonumbered files stored.

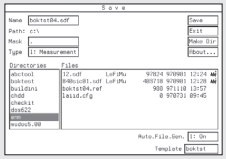
If you want to initialise the analyser with a setup file stored on a floppy disk and maintain the autostore function, do as follows:

1. Create and save your setup in the directory to be used for autonumbering
2. Copy (by means of MS-DOS) the setup file onto the floppy disk. Remember to name it *840.cfg*.

Storing a Setup...

❶ Set up the analyser as required. Remember to include display setups, noise generator setup, I/O and other settings applicable to your situation. If setup for more than one mode is to be stored, be sure to set up all modes

❷



Press COPY HDD and select the directory in which the setup shall be stored

❸

Key in the setup's file name

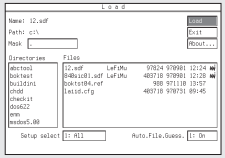
❹

Set *Type* to *Cur* to store the setup of the current mode only, or to *All* to store the setups of all the modes.

❺ Press ENTER twice or move to *Save* and then press ENTER to save the file

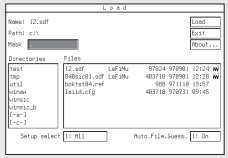
Loading a Stored Setup...

❶



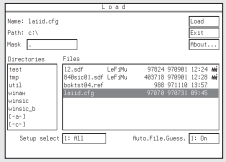
Press HDD to produce the Load menu

❷



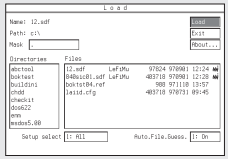
Set the mask to *.cfg*, if required

❸



Select the setup file

❹



Select *Load* and press ENTER or press ENTER on the file in the *Files* scrolling list window

A setup file is retrieved in the same way as a measurement file. However, its extension is always *.cfg*.

Set the mask to *.cfg to mask out all other files than the configuration files, if needed.

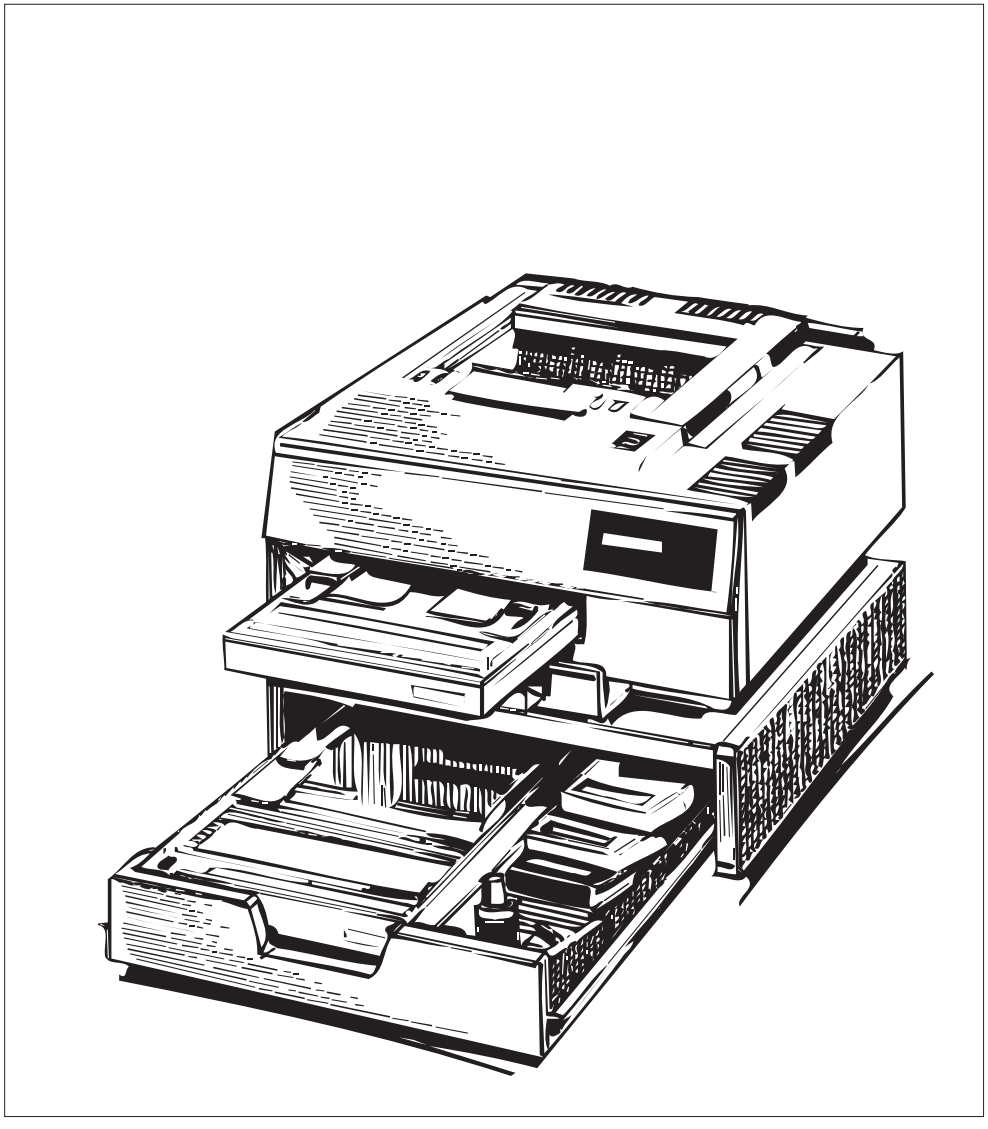
Observe the ability to personalise your analyser upon start-up by having the setup file stored on a floppy disk (the name of the setup file must be *840.cfg*) – see note below and the article on the left page of this page spread.

The configuration file will affect the instrument, measurement as well as the display setups of the modes (one or all of them) that were specified when the setup file was generated .

Note: If a setup file is stored on a floppy disk under the name of *840.cfg* and this floppy disk is inserted in the disk drive before the analyser is switched on, the analyser will pick up this setup and configure itself accordingly during start-up.

Chapter 9

116 Hardcopy Fundamentals
117 Making Screendumps
118 Making Numerical Printouts
119 Exporting Data for Spreadsheet Use



Making Hardcopies

Hardcopies or printouts are available in two formats – as graphical (i.e. screendumps) or numerical (i.e. a tabulation of values). In addition, you may choose to either direct the output to a printer or to a file.

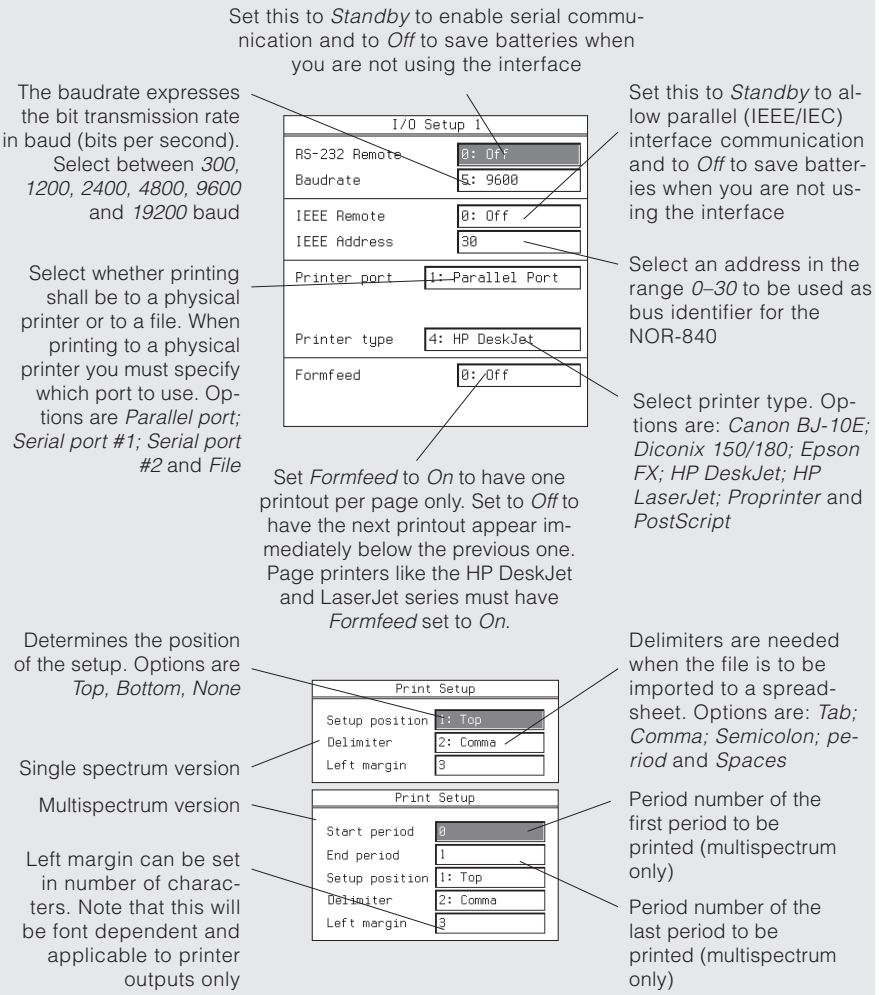
There are two menus needed for this; the I/O Setup I menu which controls to which port the printout is to be directed, the baud rate etc. and the print setup menu which allows you to specify numerical printout details.

The graphical outputs are available as screendumps only, thus no setup menu will be needed for these.

To access the I/O Setup I menu press the I/O key.

To access the Print Setup menu press 2ND PRINT.

The Menus for this Task


Set this to *Standby* to enable serial communication and to *Off* to save batteries when you are not using the interface
The baudrate expresses the bit transmission rate in baud (bits per second). Select between *300*, *1200*, *2400*, *4800*, *9600* and *19200* baud
Set this to *Standby* to allow parallel (IEEE/IEC) interface communication and to *Off* to save batteries when you are not using the interface
I/O Setup 1
RS-232 Remote
0: Off
Baudrate
5: 9600
IEEE Remote
0: Off
IEEE Address
30
Printer port
1: Parallel Port
Printer type
4: HP DeskJet
Formfeed
0: Off
Select whether printing shall be to a physical printer or to a file. When printing to a physical printer you must specify which port to use. Options are *Parallel port; Serial port #1; Serial port #2* and *File*
Select an address in the range *0–30* to be used as bus identifier for the NOR-840
Select printer type. Options are: *Canon BJ-10E; Diconix 150/180; Epson FX; HP DeskJet; HP LaserJet; Proprinter* and *PostScript*
Set *Formfeed* to *On* to have one printout per page only. Set to *Off* to have the next printout appear immediately below the previous one. Page printers like the HP DeskJet and LaserJet series must have *Formfeed* set to *On*.
Determines the position of the setup. Options are *Top*, *Bottom*, *None*
Print Setup
Setup position
1: Top
Delimiter
2: Comma
Left margin
3
Delimiters are needed when the file is to be imported to a spreadsheet. Options are: *Tab; Comma; Semicolon; period* and *Spaces*
Single spectrum version
Multispectrum version
Print Setup
Start period
0
End period
1
Setup position
1: Top
Delimiter
2: Comma
Left margin
3
Period number of the first period to be printed (multispectrum only)
Left margin can be set in number of characters. Note that this will be font dependent and applicable to printer outputs only
Period number of the last period to be printed (multispectrum only)

Making a Screendumps

Graphical printouts are available in the form of screendumps (screenshots) output to a printer or to file for later import to other software programs.

The generated file will be in PCX format, which is a format recognised by most PC software programs. Generated files will be saved in C:\ (the root directory) and their names will be 840_XXXX.PCX, in which X is a number 0~9.

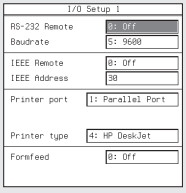
If no such file existed in advance, the first file will be named 840_0000.PCX, the next will be named 840_0001.PCX and so on.

You may have removed some of the PCX files that you have made earlier. Free "slots" will then be available and they *will* be used when storing new PCX files. However, existing files will not be overwritten.

Use MS-DOS commands to move the file(s) to a more suitable location.

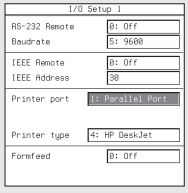
Making a Screendump

❶



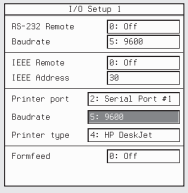
Press the I/O key

❷



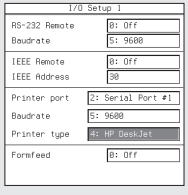
Set Printer port to the printer's port

❸



Set the baud rate, if needed

❹

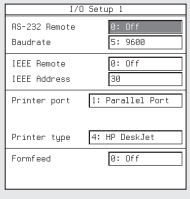


Set the printer type

❺ Press PLOT to produce a screen dump. Screen dumps always use form feed on, irrespective of your setting.

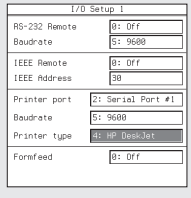
Making a PCX File of the Screendump

❶



Press the I/O key

❷



Set *Printer port* to *File*

❸ Press PLOT to generate the PCX file

Numerical, or tabulated printouts are generated in a way quite similar to the way screendumps are made. However, a few differences exist:

- A numerical printout will be made of the data associated with the active window only
- The printout will be of parameters set to *On* in the numerical table
- Only the activated spectral weighting functions wil be printed.

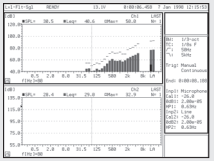
Consider the Formfeed

The formfeed function is activated from the I/O I Setup menu (press the I/O key).

When active, every new printout will appear on a new page. Page printers like the HP DeskJet and LaserJet series *must* have the form feed activated. Otherwise the paper may get stuck or some lines may not print.

When the formfeed function has been deactivated, a new printout will appear immediately below the previous.

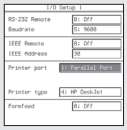
Making Numerical Printouts

❶ 

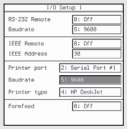
Set as active window the window whose data are to be exported

❷

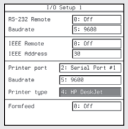
In the Display setup menu set the *Displayed columns…* to display the functions needed for your numerical printout

❸ 

Set Printer port to the printer's port

❹ 

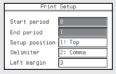
Set the *Baud rate*, if needed

❺ 

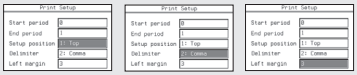
Set the *Printer type*

❻

Set *Formfeed On* or *Off* and the number of lines per page. Exit the menu

❼ 

Define the period interval (multi-spectrum only)

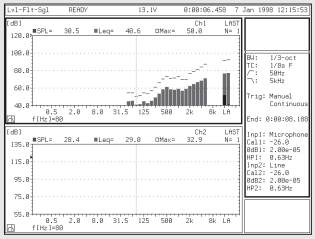
❽ 

Define the setup position, type of delimiter and the left margin. Exit the menu

❾ Press PRINT to generate the numerical printout.

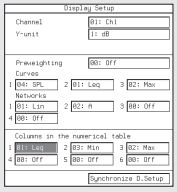
Exporting Data for Spreadsheet Use

❶



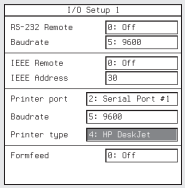
Set as active window the window whose data are to be exported

❷



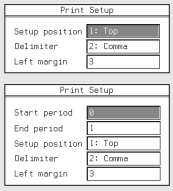
In the Display setup menu set the *Displayed columns…* to display the functions needed for your spreadsheet

❸



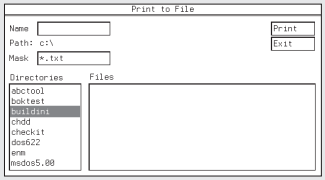
In the I/O setup menu I, set *Printer port* to *File*

❹



In the Print setup menu, define the period interval (multi…), setup position and delimiter

❺

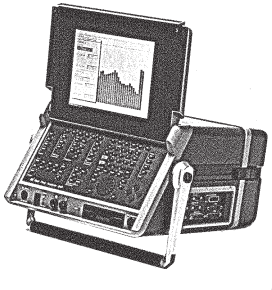


Press PRINT to enter the Print-to–file menu. Select directory and key in the file name. Select *Print* and press ENTER.

Your measurements may be exported for use in spreadsheet programs or wordprocessors and page layout programs. However, the data needs to be exported in a format readable for these programs.

Some applications tends to prefer certain file name extensions to be able to import the data correctly. The exported data are in ASCII format, making the extension .txt the obvious choice for many applications.

Although modern versions of the most popular spreadsheet programs are quite tolerant, there may still be programs demanding certain delimiters to be able to sort the data correctly. Many table editors still seem to be without options on this. Therefore the NOR-840 offers the possibility to specify the type of delimiter. You may choose between tabulator, comma, semicolon and space.



Front Panel Keys

The following contains a list of all the front panel keys in alphabetic order.

1&2. (2nd Avrg)

Combine a dual channel measurement into a single channel measurement. The function works on the contents of the Average register only. What it actually does is that it merges the contents of the two channels and puts the result of the merging into channel 1, leaving channel 2 empty. The merging is an averaging on energy basis.

1↔2. (2nd Last)

Swap the contents of the two measuring channels.

10^3

Not supported in this version.

10^{-3}

Not supported in this version.

2nd

Use this key to access all functions printed in orange colour on the front panel. When the keyboard is set to alpha mode (for example when keying in a measurement title) the 2ND key works as the SHIFT key on a PC keyboard (to key in capital letters).

3D (2nd Lf/Lt)

Three-dimensional display of multispectrum measurements.

Align (2nd Ref)

Align the cursors of the upper and lower display windows.

Alpha

Default On whenever appropriate.

Alt

The same way as the same MS-DOS key is used, i.e. it requires the MS-DOS extension to work.

Analyse

Enter the analysis part of the Nor-840 as opposed to the DOS key which is used to enter the MS-DOS mode (applies to units equipped with the MS-DOS extension only).

Auto (2nd Gain1)(2nd Gain2)

Perform input autoranging as an alternative to setting the full scale deflection manually. What it does is that it sets the full scale deflection to the maximum level available (closely connected with your calibration setting) and then measures the linear level (which is a separate "measuring channel" independent of your frequency range setting) twenty times.

It then picks out the maximum level recorded, adds 5dB and then sets the full scale deflection to the nearest legal value above this value. It must be set separately for each channel.

This means that you do not have to have the full scale deflection of both channel 1 and channel 2 set by means of autoranging. You may just as well have one set by autoranging and the other set manually.

Autoseq

Not supported in this version.

Aux

As one of the four independent (volatile) registers of the Nor-840. All files retrieved from disk end up in this register. When you retrieve a file, you will overwrite the current contents of the Aux register.

There is one Aux register for each measurement mode. Hence, the Aux register of the multi-spectrum mode is not the same register as the Aux register of the single-spectrum mode or the reverberation mode (optional).

However, they all share a common memory-pool. Which means that if one of the Aux registers contains a large amount of data, this may affect the amount of free memory available for the Aux registers in the other modes.

Av\La (2nd Comb)

As an undo function when you have merged a data set contained in the Last register with the contents of the Average register. It should be read as Average less Last.

There is only one level of undo, i.e. you cannot repeatedly hit the Av\La to undo previous mergings.

Avrg

As one of the four independent (volatile) registers of the Nor-840. Merging (in the sense of averaging on energy basis) can only

take place in this register. To merge two sets of data, place one in the Average register and copy the other into this register or use the Combine function.

To avoid unintended erasure of data, the copy and combine functions have been made identical for the Average register.

Cal

Enter the Calibration menu of the Nor-840.

Clear

Clear the contents of a register.

Comb.

Merge the dataset contained in one register with a dataset contained in another.

When a dataset is combined with the contents of the Average register (with the Average register as destination), the merging will be an energy based averaging of the levels of the overlapping frequency bands while at the same time retaining the levels of the non-overlapping frequency bands. Single-spectrum only!

Cont

Resume a paused or halted measurement. If the measurement was interrupted before it terminated, hitting CONT will cause the measurement to be resumed and go on until the Measurement end condition has been met.

If the measurement terminated by itself, hitting CONT will cause the instrument to prolong the measurement period and go on measuring until the Measurement end condition once again has been met.

Copy

Copy the contents of a register to another register, leaving the source register contents intact. The contents of the destination register will be overwritten (except when the destination is the Average register)

Ctrl

The same as the corresponding key on an MS-DOS PC.

Cursor

Select the graph cursor as the function controlled by the DIAL and the PREV & NEXT keys.

D.Setup

Produce the Display setup menu. Note that there is a separate display setup menu for each measuring mode.

Del

Delete characters to the right of cursor whilst editing in a string field.

Dial

Scroll through the valid states or values of a parameter, including the position of the graph cursor. Also used in scroll lists.

When applied to cursor and graph functions, the function to be controlled (for example Y-range) must be selected first.

Disk

Store or retrieve data and set-ups to and from the floppy disk.

DOS

Enter the MS-DOS mode and run the Nor-840 as an IBM compatible PC. Requires MS-DOS extension installed.

Edit

In numerical tables values may be edited by the user. Edit is used to enter this function.

End

Move to the end of a menu, however, not inside a list field. Inside a list field it is used to go to the end of the list.

Enter

Confirm settings and selections, as well as exit to menus.

Esc

As the corresponding MS-DOS key, but also to leave a menu ignoring (undoing) all changes made to the menu.

F-keys

As the corresponding MS-DOS keys. F12 displays information about the Nor-840 software version and stops the hard disk.

FFT

Enter FFT mode.

Field Selector keys

Select a parameter field inside a menu.

Filter

Select fractional octave bands as measurement mode (as opposed to FFT)

Gain1/Gain2

Produce the menu for setting the full scale deflection of the corresponding measurement channel.

Gen

Switches the built-in noise generator ON/OFF.

Harm

In FFT mode, use this function to reveal any harmonic relations in the signal investigated.

HDD

Store or retrieve data and set-ups to and from the (optional) hard disk, but can also be used to store and retrieve data to and from the floppy disk.

Help

Activate the Help function.

Home

Move to the beginning of a menu, however, not inside a string field. Inside a string field it is used to go to the beginning of the string, word, value as well as the top line inside a list box (such as the files list in the Save menu).

I/O

Access the I/O set-up menu, controlling such things as remote control ports, baud rate, IEEE address, printer port and printer type.

Index

List the paths to where user-defined weighting networks are stored.

Input

Produce the Input source menu.

Insert

Toggle between Insert and Overwrite in a string field.

Integr

Not supported in this version.

Intens

Entrance key to intensity mode.

Last

As one of the four independent (volatile) registers of the Nor-840. Measured data end up in this register. Data can be transferred from but not to this register.

Level

Select level (as opposed to sound intensity) as measurement mode.

Lf/Lt

Select between spectrum and time history profile of a multi-spectrum mode measurement.

M.Setup

Access the Measurement setup menu. Note that there is a separate menu for each measurement mode.

Move

Move data from one register to another. Note that the source register will be empty and destination register contents will be overwritten once this command has been executed.

Multi

Select Multi- (multiple spectra) as opposed to Single (one spectrum only) as measurement mode.

Next

Scroll through a parameter's values or the cursor's valid positions in the smallest steps available towards increasing values.

Num

Produce a tabulation of acquired or retrieved values (as opposed to displaying a graph).

Page Up/Page Dn

Move page-wise up & down in a scroll list.

Pause

Temporarily halt an ongoing measurement. Use CONT to resume.

Plot

Produce a screendump which is written to the selected device (printer, disk)

Prev

Scroll through a parameter's values or the cursor's valid positions in the smallest steps available towards decreasing values.

Print

Produce numerical printouts.

Record

Not supported in this version.

Ref

Used to visualise the difference between a level at one point of a graph and another point at the same graph. For spectra, the display will indicate level difference and frequency difference, for time profiles (level vs. time) the display will indicate level difference and time difference.

Also used to insert a spectrum into the reference/spectral weighting list (the W1~W8).

RT

Calculate the reverberation time based on a multispectrum measurement.

Setup (2nd Autoseq)

Not supported in this version

Setup (2nd Help)

Set the HELP level (the amount of warnings you get when data are about to be jeopardised)

Setup (2nd Record)

Not supported in this version.

Setup (2nd Gen)

Set-up the noise generator, such as the noise type, output level etc.

Setup (2nd Plot)

Not supported in this version.

Setup (2nd RT)

Set up the basis for the reverberation time calculation, such as type of excitation, distance to background noise etc.

Setup (2nd Analyse)

Select Master Instrument Mode, i.e. normal or MLS mode (applies to units with MLS extension installed only).

Setup (2nd User)

Set real time clock. Clock will be set to indicated time when Enter is pressed.

Setup (2nd Print)

Define variables for the numerical print-out, such as Start & End period (Multispectrum only), margin and data delimiter.

Single

Select Single (one spectrum only) as opposed to Multi- (multiple spectra) as measurement mode.

Size

Not supported in this version.

Start

Start a measurement. The data acquisition will not start until the preset trigger condition has been met.

Stop

Stop an ongoing measurement. Resume by pressing CONT.

Tab

Applies to MS-DOS mode.

Title

Key in a measurement title, up to 10 lines @ 40 characters each.

Trig

Access the Trigger condition menu.

Type

Switch between different display modes.

User

Provides access to the User register.

X-min

Select the left-most point of the horizontal axis as the function controlled by the DIAL and the PREV & NEXT keys.

Y-max

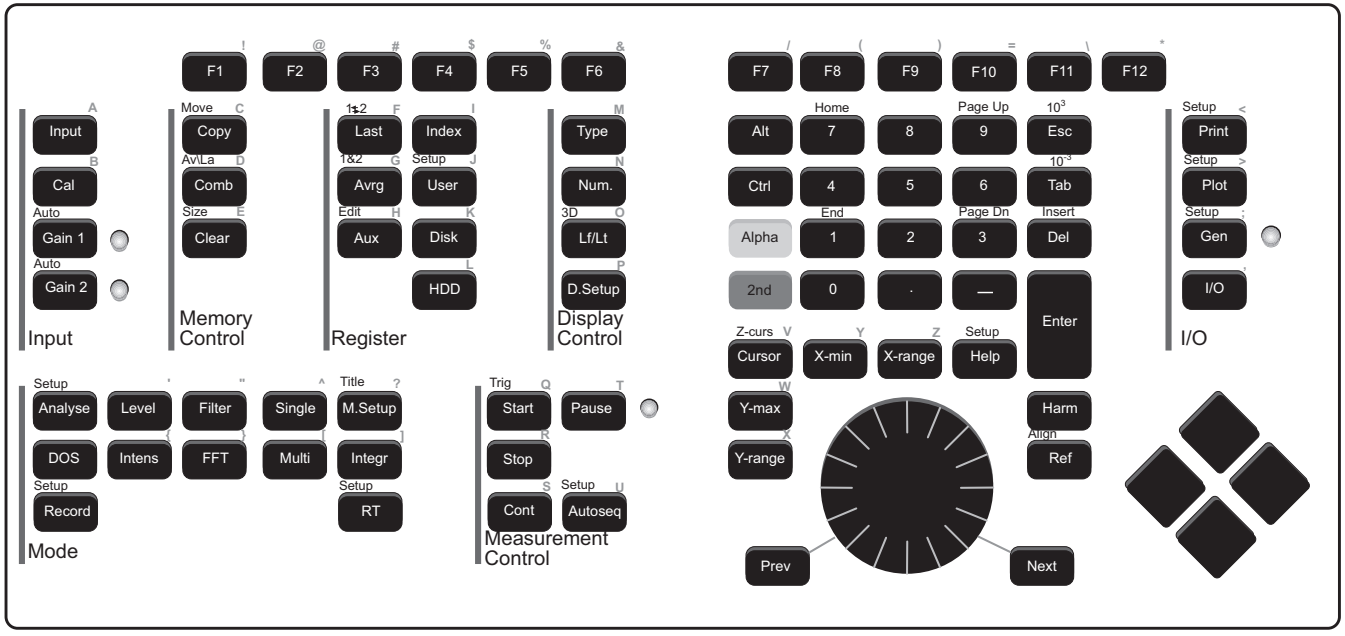
Select the vertical axis top scale value as the function controlled by the DIAL and the PREV & NEXT keys.

Y-range

Select the vertical axis range as the function controlled by the DIAL and the PREV & NEXT keys.

Z-cursor (2nd Cursor)

Select the graph cursor of the time axis when displaying the frequency spectrum and of the frequency axis when displaying a time history profile. The selection will be a function controlled by the DIAL and the PREV & NEXT keys. Applies to multispectrum mode only.



Technical Specifications

The below specifications apply to all new models. Due to continuous technological advances, older versions of the analyser may have specifications that are slightly inferior. In particular, the storage capacity (the hard disk size), the battery capacity and the analyser's ability to suppress any effects of mechanical shocks have been significantly improved on newer units.

ANALOGUE INPUTS

Number of channels: Two
(Nor-840 is also available in single channel version.)
Microphone inputs (two): 7-pin LEMO connectors, B&K type JJ0723 on request.
Preamplifier voltage: 120 V, 3 mA to each preamplifier.
Polarisation voltage: 0, 28 or 200V selectable, ±1%.
Direct inputs (two): BNC connectors.
Charge inputs (two): TNC connectors.
Intensity Input: 18-pin LEMO connector.
Maximum input signals: ±120 V_{PEAK} (all inputs except charge).
Input impedance:1 MΩ/200 pF (all inputs).

INPUT AMPLIFIERS

Amplifier gain: –40dB to +70dB in 5dB steps. Additional gain:0–10 dB with accuracy and resolution 0.1 dB for calibration purposes.
Measurement range: 0.03mV–100Vrms. Corresponds to SPL values from –30dB to +160dB with a microphone sensitivity of 50mV/Pa.
Amplification error: Max. 0.2dB (20Hz–12.5kHz)
Frequency range (AC output): 0.1Hz–50kHz within ±0.5dB. 0.05Hz–100kHz within ±3dB.
High-Pass filters (–3dB): 3-pole Butterworth set at 0.5Hz or 16Hz (all inputs).
Low-Pass filter (–3dB): 3-pole Butterworth selectable at 2 kHz (charge inputs only).
Hum and Noise, Microphone inputs: Measured with preamplifier type 1201 and 18 pF microphone equivalent. FSD 50dB, calibration: –26.0dB re. 1V.
HP filter set to 0.5Hz
0.8–5Hz: <20dB
6.3–20Hz: <12dB
25–100Hz: <7dB
125–315Hz: <0dB
400–3150Hz: <–4dB
4–20kHz: <0dB
Lin network: <15dB
A network: <10dB
Hum and Noise, Line inputs: Measured with Line input short-circuited to ground, FSD 50dB, HP filter set to OFF, calibration –26.0dB re. 1V
0.1–25Hz: <–30dB
31.5–100Hz: <–25dB
125–400Hz: <–20dB
500–1600Hz: <–15dB
2–5kHz: <–10dB
6.3–20kHz: <–5dB
Lin network: <+3dB
A network: <0dB
Hum and Noise, Charge inputs: Measured with a1nF capacitor connected to the Charge input. HP filter set to OFF. Calibration –240dB re. 1V implying that 180dB ≡1nC. FSD 140dB.
0.1–250Hz: $<5.6\times10^{-16}$ C
315–1250Hz: $<7\times10^{-16}$ C
1.6–10kHz: $<18\times10^{-16}$ C
12.5–20kHz: $<32\times10^{-16}$ C
Lin network: $<100\times10^{-16}$ C
A network: $<100\times10^{-16}$ C

ANALOGUE OUTPUTS

Wideband outputs (two): BNC connectors. The output signal comes directly from the input amplifiers.
Output level: ±10Vpeak, 1.0Vrms corresponds to full scale deflection on display. Outputs are short-circuit proof to ground and output current is in excess of 10 mA.
Output impedance: Max. 10Ω.

ANALOGUE-TO–DIGITAL CONVERTER

Converter type: Sigma delta with 64 times oversampling
Sampling rate: 64 kHz (15.625 msec).

FILTERS AND NETWORKS

Anti-aliasing filter: Combined analogue and digital low-pass filter.
Passband ripple: <0.1 dB.
Stopband attenuation: >75 dB above 1.3 × cut-off frequency.
Phase matching error: < 0.1° for 10Hz–5kHz. <1.0° for 0.1Hz–20kHz.
Digital filters: 6-pole IIR filters for 1/1- and 1/3-octave bands. The 1/3-octave centre frequencies are set with the factor $10^{n/3}$.
Frequency range (dual channel): 0.125–16000Hz for 1/1-octave bands (centre frequency). 0.1–20000Hz for 1/3-octave bands (centre frequency).
Filter response: The 1/1- and 1/3-octave filters meet the requirements from IEC-

61260 class 0, ANSI S 1.11 - 1986 Type 1D order III
Weighting networks: The true A- and Lin-networks fulfill the requirements of IEC 60651 Type 0 and ANSI S 1.4 1983 Type 0 for precision sound level meters.
Lin-netw. response: 25–16 000Hz (–0.3dB) 17.8–22 300Hz (–3dB)
Calculated networks: A-, B-, C- and Lin-networks as well as up to four user-defined networks may be calculated as a sum of the frequency bands within the selected frequency limitations.

SOUND INTENSITY

Compliance: The sound intensity option complies with IEC 61043 class 1 and ANSI S1.9-1996 class 1

LEVEL DETECTOR

Detector type: Digital true RMS.
Resolution: 0.1dB in accordance with IEC 60651 and IEC 60804 Type 0, as well as ANSI S 1.4 - 1983 Type 0.
Reference range: 40–120dB in accordance with IEC 60651/60804 and ANSI S 1.4 - 1983
Primary indicator range: 50–120dB with CF _ 3 (50–110 at CF_10) in accordance with IEC 60651 type 0 and ANSI S 1.4–1983 type 0.
Linearity range: 75 dB in accordance with IEC 60804
type 0.
Pulse range: 78 dB in accordance with IEC 60804 type 0.
Reference frequency: 1000Hz
Reference SPL: 114.0dB SPL
Accuracy (20Hz to 12.5kHz) measured at reference range:
+5dB above to 60 dB below FSD: <0.2 dB
60 dB to 70 dB below FSD: <0.4dB
70 dB to 80 dB below FSD: <1.0dB
Time constants: Selectable in a binary sequence from 1/16 sec. to 8 sec. plus I(mpluse). The 1/8 sec as F(ast), the 1 sec. as S(low) and the I(mpulse) are in accordance with IEC 60651 Type 0 and ANSI S 1.4 - 1983 Type 0.
Integration period: 4 msec. to 100 hours with 1msec. resolution
Crest factor capability: 10 dB crest factor margin at FSD increasing to 90 dB at 80 dB below FSD
Overload detector: The overload detector operates in accordance with IEC 60651, IEC 60804 and ANSI S 1.4 1983 standards. The overload detection starts at 10dB (PEAK) above FSD

DISPLAY

Display type: 10" bright, backlit, double twisted, monochrome LCD screen or (optional) back-lit 10.4"VGA colour screen.
Resolution: 640 (horiz.) x 480 (vert.) pixels (VGA standard).
Display formats: One or two windows, with or without annotation and setup information. Each window may individually present data from selectable channel and selectable parameter(s), as a numerical table, as a level vs. frequency graph, or as a level vs. time graph.
Displayed level range: 20, 40, or 80dB. User selectable.
Graduation: 0.1dB–0.8dB depending on selected format.
Numeric graduation: 0.1dB.
Numerical range: –99.9dB to +199.9dB.

MEMORY

Internal storage: Up to 10.000 full frequency single channel spectra. Depends on No. of channels employed as well as registers, parameters and frequency range range used. Volatile memory.
Floppy-disk: 1.44 Mbyte 3½" floppy-drive (MS-DOS compatible format). Capacity approximately 250 single-spectrum measurements or 100 multi-spectrum measurements @1000 periods each.
Hard disk (optional):2.1 Gbyte hard disk (MS-DOS compatible format). Capacity approximately 391 000 single spectrum measurements or 153 000 multispectrum measurements @1000 periods each.

DIGITAL INTERFACES

Control: Almost any setting or any data read-out may be made using the digital interfaces. See also *Remote Control Commands* (separate booklet).
IEEE-488: Meets the IEEE Standard 488-1978.
RS-232C (three): Meets the RS-232C Standard for normal serial interfaces with handshakes.
Printer: Graphic screen-dumps or numeric tables may also be printed out using the Centronics parallel interface.

SIGNAL GENERATOR

Output: BNC connector.
Output impedance: <10Ω (±10mA).
Signal types: Random or Pseudo-Random noise.
Spectra: White, Pink, 1/1-octave or 1/3-octave noise.

Repetition rate: Approximately 28 minutes corresponding to 0.00006Hz spectral line separation.
Filters: The 1/1- and 1/3-octave filters meets the requirements from IEC-1260 class 1 and ANSI S 1.11 - 1986 Type 1D.
Output level: Selectable in 1dB steps in the range 0 to –60 dB re. 1VRMS.

GENERAL

Power requirements: 30W, 11–15 VDC
Dimensions: 34×21×35 [cm], 13.4×8.3×13.8 [inches] (W×H×D) without battery case.
Weight: 9.7 kg /21.4 lb. without batteries (12.3 kg /27.1 lb. with Battery Case Nor-330A).
Warm up time: <30sec for 0.1dB accuracy. When used with condenser microphones, no calibration should take place until polarisation voltage has settled (approximately 2 minutes).
Enclosure class: IP20, IP40 (with closed front cover).

ENVIRONMENTAL

Temperature range: Storage: –20 to +70 °C, gradient 15 °C/hour (–4 to +158 F, gradient 59 F/hour).
Operating:+5 to +55 °C, gradient 15 °C/hour (41 to 131 F, gradient 59 F/hour)
Temperature drift: <0.002 dB/°C (without microphone)
Relative humidity: 5–90% (provided no condensation) for storage and 8–80% for operation.
Vibration: 5m/s² (operation) 50m/s² (storage).
Shock: 100G (11msec half sine wave) for operation and 200 G for storage/transport.

OVERALL PERFORMANCE

The overall performance of the Nor-840 with a suitable microphone and preamplifier, such as the Nor-1220/1225/1230 and Nor-1201, corresponds to the Sound Meter Level Standards IEC 60651 Type 1, IEC 60804 Type 1, and ANSI S 1.4 - 1983 Type 1. (Type 0 with suitable microphones)

MAINS ADAPTOR Nor-329A

Mains input: 93–130 VAC (US version) or 198–250 VAC 50/60 Hz
Output: 13.2 VDC/135 WMAX
HF noise at input and output: Approved in accordance with CISPR-14
Protection: Short circuit proof, thermal protection, protected against over and undervoltage.
Enclosure class: IP 20
Dimensions: 192×109×53.5 [mm], 7.6×4.3×2.1 [inches]
Weight: 0.85kg/1.9 lb.

BATTERY PACK Nor-330A

Battery capacity: 5Ah, gives approximately 2.5 hours of continuous operation with two microphones/preamplifiers.
Charging time: 2 hours
Weight: 3.3 kg/7.3 lb.

BATTERY PACK Nor-332

Battery capacity: 10Ah, gives more than 7 hours of contiuous operation with two microphones/preamplifiers.
Charging time: 16 hours
Weight: 6.2 kg/13.7 lb.

All specifications subject to change without further notice.

Declaration of Conformity

We, Norsonic AS, Gunnersbråtan 2, Tranby, Norway, declare under our sole responsibility that the product:

Real Time Analyser NOR-840

FROM SERIAL NUMBER 18711

including mains adaptor 329A, battery pack 330A or 332A, to which this declaration relates, is in conformity with the following standards or other normative documents:

Performance complying with:

IEC 60651 type 0
IEC 60804 type 0
IEC 61043 class 1
IEC 61260 class 0
ANSI S1.4-1983 type 0
ANSI S1.11-1986 order 3 type 1D
ANSI S1.9-1996 class 1

EMC: EN 50081-1
EN 50082-1

Safety: EN 61010-1993 for portable equipment and pollution category 2

This product has been manufactured in compliance with the provisions of the relevant internal Norsonic production standards. All our products are tested individually before they leave the factory. Calibrated equipment—traceable to national and international standards—has been used to carry out these tests.

This Declaration of Conformity does not affect our warranty obligations.

Tranby, May 1998

The declaration of conformity is given according to EN 45014 and ISO/IEC Guide 22.

Norsonic AS, P.O. Box 24, N-3420 Lierskogen, Norway

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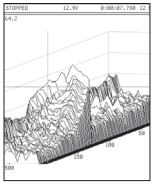

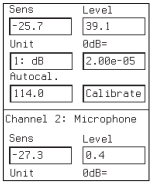

Book Level

Beginning
- ✓ **Some experience**
- ✓ **Intermediate**
- ✓ **Advanced**

Tutorial
- ✓ **How-to**
- ✓ **Reference**

Completely Revised, Expanded & Up-to-Date

Your approach to the Nor-840 documentation depends on what you want to do and how much you already know. The User Documentation has been designed to help you get more benefits from all the analyser's features in less time than ever before.

Need a quick start guidance? Read the Getting Started section! This part of the manual outlines, in just a few pages, all the fundamentals needed to start using the analyser – from the hardware of battery handling to the software of menu handling!

Need in-depth knowledge about a certain topic? The concept provides detailed information at a glance! All the related information is compiled and presented on a single page or a page spread!

Need to know the correct procedure for setting the analyser up? The order of appearance of the topics reflects the recommended sequence.

Looking for certain topic? The extensive index provides the keywords you need!

Norsonic AS supplies a complete range of instrumentation for acoustics – from sound calibrators, microphones & preamplifers; via small handheld sound level meters to advanced, yet portable, real time analysers, but also spectrum shapers, building acoustics analysers and complete community, industry and airport noise monitoring systems. Contact your local representative or the factory for information on our complete range of instrumentation.

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